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THE ACOUSTICAL SOCIETY OF AMERICA

Founded 1929

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NOTICE OF CHICAGO MEETING

The next meeting of the Society is to be held in Chicago on December 13th and 14th. The meeting-place will be Lincoln Hall of the Northwestern University School of Law on McKinlock Campus.

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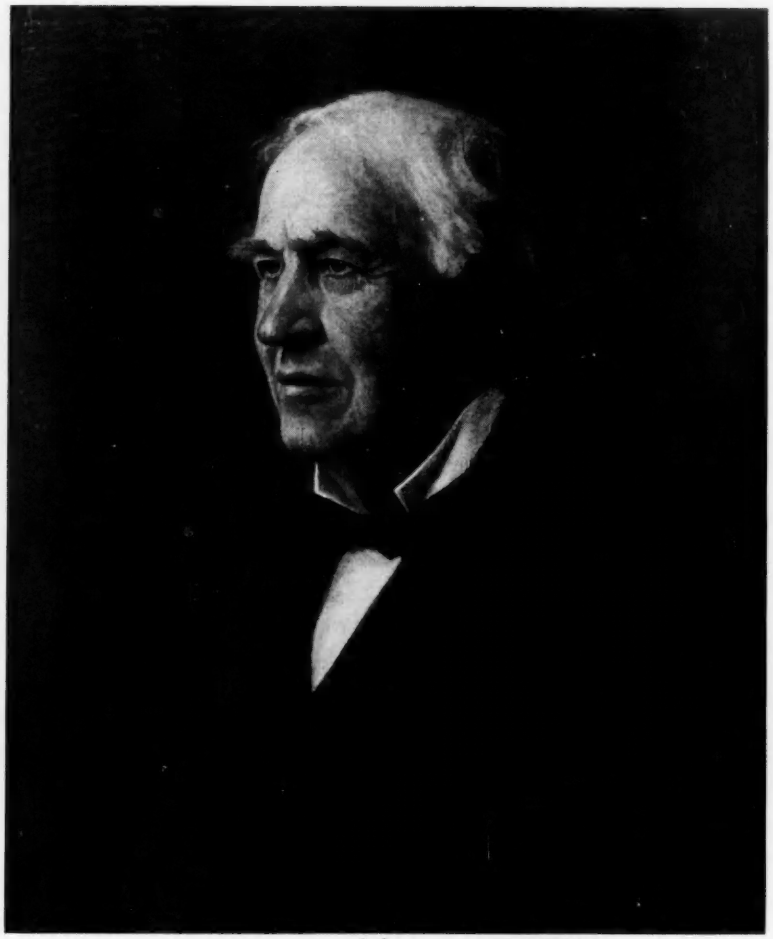

Thos A Edison

Physics
Faxon

FIFTY YEARS AGO Thomas Alva Edison made the first practical electric light and this year the world is paying tribute to him in grateful recognition of the benefits to civilization which have been brought about as a result of his achievement.

Mr. Edison's genius has made its impression on many fields of research and his contributions to acoustics have been of major importance. In recognition of this the Acoustical Society of America, by unanimous vote of its Executive Council, has chosen him to be the first Honorary Member of the Society.

In acceptance, Mr. Edison has written the following letter:

FROM THE LABORATORY
OF
THOMAS A. EDISON
ORANGE, N. J.

June 17, 1929

Mr. Harvey Fletcher, President,
Acoustical Society of America,
463 West Street,
New York City

Dear Mr. Fletcher:

Your letter of May 21st was forwarded to me at Fort Myers, Florida, and I assure you that I feel highly gratified and honored that the Acoustical Society of America have, by unanimous vote of the Executive Council, chosen me to be the first honorary member.

Naturally, I am greatly pleased to be offered this place of honor and I am very glad indeed to accept the membership which you have so kindly offered me.

Yours very truly,

THOS. A. EDISON

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THE JOURNAL
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Volume I Number 1

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HISTORY OF ACOUSTICAL SOCIETY OF AMERICA

By Wallace Waterfall

(As read at the beginning of the business meeting on May 10th)

It is usually difficult to assign a definite day or place to the conception of the idea which leads to the formation of an organization. Subsequent developments of the idea are more definite. In the following paragraphs I am giving a brief account of the happenings which have lead to the present organization of the Acoustical Society of America.

During the past five years my activities have brought me in contact with a large number of professional men engaged in experimental work in connection with architectural acoustics. There was a noticeable lack of coördination of activities. For greater development of the subject these professional men needed a more intimate knowledge of the rapidly growing practical requirements in architectural acoustics and there was no question but what the public would benefit by a more complete knowledge of the work being done in the laboratories. There were many differences of opinion on fundamental points which should be submitted to discussion for proper solution. These conditions in the field of architectural acoustics made the need of some sort of organization quite obvious and from time to time I discussed the possibility of this with many professional men I met.

The idea took its first definite form during the latter part of the summer of 1928, when Professor F. R. Watson and I happened to be on the West Coast at the same time. I am not exactly sure of the date but believe it was one day during the first week of September that Professor Vern O. Knudsen invited Professor Watson and me to accompany

him to his Beach Club near Los Angeles for dinner one evening. During the evening we discussed the formation of a society and it was definitely agreed that letters were to be sent to a group of about fifteen men who had been contacted regarding such an organization. Professor Watson and Professor Knudsen agreed to supplement my letter to these men with a letter of their own regarding the desirability of the organization.

My first letter was sent to the following group of men.

Professor F. R. Watson
Professor A. L. Foley
Professor G. R. Anderson
Professor F. A. Saunders
Professor D. L. Rich
Mr. C. M. Swan
Professor W. R. Barss
Mr. V. L. Chrisler
Dr. Paul R. Heyl
Dr. Harvey Fletcher
Professor Dayton C. Miller
Dr. Paul E. Sabine
Professor V. O. Knudsen

Replies were not received from all of these men but the letters which were received expressed interest and added some new ideas to the program. Up to that time we had considered the organization more as a society of acoustical engineers and had thought chiefly of architectural acoustics. Professor Miller pointed out the desirability of enlarging the scope of the society so as to make it a more stable organization. At about the same time Professor Knudsen and Dr. Fletcher advanced the same idea.

Enlarging the organization to include workers in all branches of acoustics brought many new possibilities. I happened to be in New York about this time and had quite a talk with Dr. Fletcher. He suggested that it might be advisable to get a group together during the meetings of the American Association for the Advancement of Science. He offered the use of the Bell Telephone Laboratories as a meeting place and between us December 27th was chosen as the date of the meeting. On December 10th letters were sent to all of the men in the previous list and in addition, to Mr. R. V. Parsons, Mr. M. A. Smith and Mr. B. J. Graham. In this letter to these sixteen men they were told of the date of the meeting and the fact was mentioned that the proposed organization had been enlarged in scope to include workers in all branches of acoustics. It was mentioned that there were many men who might be interested in such an organization and each person who received the letter was asked to take it upon himself to write as many persons as he thought might attend the first meeting or be interested in the society. Most everyone cooperated and the result was the meeting on December 27th at the Bell Telephone Laboratories.

At that meeting the following men were in attendance.

G. R. Anderson, University of Toronto
F. R. Watson, University of Illinois, Urbana, Ill.
John Bellamy Taylor, General Electric Company, Schenectady, N. Y.
Dayton C. Miller, Case School of Applied Science, Cleveland, Ohio
Harvey Fletcher, Bell Telephone Laboratories, N. Y. C.
Bert. J. Graham, Macoustic Engineering Company, Cleveland, Ohio
Charles Fuller Stoddard, American Piano Company, N. Y. C. (Research Lab.)
C. N. Hickman, American Piano Company, N. Y. C. (Research Lab.)
Horatio W. Lamson, General Radio Company, Cambridge, Mass.
V. L. Chrisler, Bureau of Standards, Washington, D. C.
H. C. Harrison, Bell Telephone Laboratories, N. Y. C.
Irving Wolff, Radio Corporation of America, VanCortlandt Pk. S. & Saxton Ave., N. Y. C.
Joseph B. Kelly, Bell Telephone Laboratories, N. Y. C.
Vern O. Knudsen, University of California at Los Angeles, Los Angeles, Calif.
E. C. Wente, Bell Telephone Laboratories, N. Y. C.
Warren C. Jones, Bell Telephone Laboratories, N. Y. C.
R. V. Parsons, Johns-Manville Corporation, N. Y. C.
Lonsdale Green, Jr., Johns-Manville Corporation, N. Y. C.
Wallace Waterfall, The Celotex Company, Chicago, Ill.
H. A. Frederick, Bell Telephone Laboratories, N. Y. C.
J. P. Maxfield, Victor Talking Machine Co., Camden, N. J.
Walter A. MacNair, Victor Talking Machine Co., Camden, N. J.
R. F. Mallina, Victor Machine Co., Camden, N. J.
D. G. Blattner, Bell Telephone Laboratories, N. Y. C.
A. T. Jones, Smith College, Northampton, Mass.
John H. Boland, U. S. Gypsum Co., 17 State St., N. Y. C.
Frederick A. Saunders, Jefferson Physical Lab., Harvard University, Cambridge, Mass.
H. D. Arnold, Bell Telephone Laboratories, N. Y. C.
E. J. Schroeter, Macoustic Engineering Co., Cleveland, Ohio
H. A. Erf, Nor. O. Tel. Co., Bellevue, Ohio
Robert H. Schroeter, Macoustic Engineering Co., Cleveland, Ohio
Donald MacKenzie, Bell Telephone Laboratories, N. Y. C.
F. K. Richtmeyer, Cornell University, Ithaca, N. Y.
L. J. Sivian, Bell Telephone Laboratories, N. Y. C.
R. L. Wegel, Bell Telephone Laboratories, N. Y. C.

N. R. French, American Telephone & Telegraph Co., N. Y. C.
E. L. Norton, Bell Telephone Laboratories, N. Y. C.
W. P. Mason, Bell Telephone Laboratories, N. Y. C.
Clarence W. Hewlett, General Electric Company, Schenectady, N. Y.
John C. Steinberg, Bell Telephone Laboratories, N. Y. C.

Dr. Harvey Fletcher was elected as chairman of the meeting and a general discussion was started in which many reasons both for and against the formation of the society were given. After the discussions were concluded a vote was taken and it was decided to form a society which should be known as the Acoustical Society of America. A temporary set of officers was elected as follows:

President, Harvey Fletcher
Vice President, V. O. Knudsen
Secretary, Wallace Waterfall
Treasurer, Charles Fuller Stoddard

All of the persons present signified their desire to become members of the Society. A committee of ten, consisting of the four temporary officers and the men named below, was appointed to draft a constitution and by-laws.

D. C. Miller	J. P. Maxfield
F. A. Saunders	R. V. Parsons
F. K. Richtmeyer	F. R. Watson

CONSTITUTION AND BY-LAWS AS ADOPTED BY THE SOCIETY, MAY 10, 1929

CONSTITUTION

Article I. Name

The name of the society shall be the "Acoustical Society of America."

Article II. Purpose

The purpose of the society shall be to increase and diffuse the knowledge of acoustics and promote its practical applications.

Article III. Membership

The society shall be composed of the subscribers to this constitution at the date of organization and such other persons or corporations as shall be hereafter admitted to membership in accordance with the provisions of the by-laws.

The membership shall be classified as fellows, honorary members, sustaining members, and members, as provided in the by-laws. Sustaining members may be either persons or corporations; all other members must be persons. The voting power shall be restricted to and vested exclusively in the members constituting the fellows of the society.

Article IV. Executive Council

The affairs of the society shall be in charge of an executive council which shall consist of the president, vice president, secretary and treasurer and six fellows to be elected as provided in the by-laws.

Article V. Officers

The officers of the society shall consist of a president, a vice president, a secretary, and a treasurer, who shall be chosen in the manner provided in the by-laws.

Article VI. Amendments

This constitution may be altered or amended at any meeting of the society when a proposed amendment, as submitted by the executive council, is favored by two-thirds of all of the fellows present thereat voting upon such amendment. A copy of every proposed amendment shall be mailed to each fellow of the society at least thirty days before a vote is to be taken thereon. All voting shall be by ballot.

This constitution may be subscribed to in any number of counterparts, all of which shall constitute one and the same instrument, irrespective of the time or times when the same, or any of the counterparts, was or may be subscribed to.

BY-LAWS

Article I. Name and Purpose of the Society

The name and purpose of this society shall be those stated in the constitution.

Article II. Membership

Section 1. Eligibility. There shall be four classes of membership viz.: honorary, fellows, sustaining members, members.

Any person or corporation interested in acoustics shall be eligible for membership in this society. Every person or corporation desiring to become a member of this society shall in person or by authorized representative sign an application for such membership.

Any person of outstanding eminence in theoretical or applied acoustics may be elected to honorary membership.

Any person who shall have rendered conspicuous service and made notable contributions in acoustics shall be eligible to fellowship in this society.

Any person or corporation contributing Fifty Dollars ($50) or more annually may be elected a sustaining member.

Section 2. Election. Applications for membership must be signed by two or more fellows or members and forwarded to the secretary. Applicants shall be admitted to membership when their applications have been approved by three or more members of the executive council.

All classes of membership other than honorary shall be elected by a majority vote at a regular or special meeting of the executive council. Honorary members shall be elected only by a unanimous vote of the executive council.

Section 3. Privileges. The right to vote at elections, and hold office shall be restricted to and exclusively vested in the membership constituting fellows. All other classes of membership, other than honorary, shall have the same privileges as fellows, except as herein otherwise provided.

Honorary members shall have such privileges as may be granted by the executive council.

Section 4. Termination of Membership. Membership in the society may be terminated at any time by a majority vote of the executive council.

Failure to pay dues for one year would be deemed sufficient cause for termination of membership.

Article III. Dues

The annual dues of the society shall be Seven Dollars and Fifty Cents ($7.50) for fellows and Five Dollars ($5.00) for members.

Disbursements shall be made by the treasurer when such disbursements are approved by a two-thirds vote of the executive council.

Article IV. Officers

The officers of the society shall consist of a president, a vice president, a secretary and a treasurer.

Section 1. Eligibility. Any fellow is eligible for election to any office in the society.

All officers except members of the council shall be elected for a period of one year.

Two members of the council shall be elected each year and shall hold office for a period of three years.*

All officers shall take office two weeks after election.

After serving two consecutive terms the president shall be ineligible for re-election. After serving one term members of the executive council shall be ineligible for immediate re-election.

Section 2. Election. Election shall be by ballot. Nominations shall be made by a nominating committee appointed by the president and approved by the council. This committee shall consist of five fellows of the society, at least one of whom shall, when present at the meeting, be a past president of the society. The name of any candidate suggested by more than ten fellows shall appear on the official ballot along with those nominated by the nominating committee.

Article V. Meetings

The meetings of the society and the executive council shall be held at such times and place as may be designated from time to time by the executive council.

Article VI. Order of Business

The order of business shall be determined by the executive council.

* At the first election two members of the council shall be elected for a term of one year, two for a term of two years, and two for a term of three years.

Article VII. Executive Council

The affairs of the society shall be managed by an executive council which shall consist of the president, vice president, secretary and treasurer and six members to be elected as provided in these by-laws. Each member of the council shall be afellow in good standing at the time of election.

Any vacancy occurring in the executive council may be filled for the unexpired term by a majority vote of the remaining members of the executive council.

Article VIII. President

The president shall, if present, preside over all meetings of the society and the executive council. In case of the absence of the president, the vice president shall have all the powers and perform all the duties of the president.

Article IX. Secretary

The secretary shall attend all meetings of the society and the executive council and shall keep a true and faithful record thereof.

Article X. Treasurer

The treasurer shall have charge of and receive all the funds and securities of the society. He shall deposit the funds to the credit of the society in such bank or trust company as the executive council shall direct and he shall disburse the same only upon the approval of the executive council.

Article XI. Amendment

Upon recommendation of the executive council, these by-laws may be altered or amended at any meeting of the society by a majority vote of the members present. Before the executive council will make any recommendations, the substance of the proposed alterations or amendments must have been submitted to each of its members.

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Vote of Appreciation to Bell Telephone Laboratories

At the beginning of the Saturday afternoon session the following vote of appreciation was given to the Bell Telephone Laboratories at the suggestion of Professor F. R. Watson:

Moved and carried that the Acoustical Society express its appreciation to the Bell Telephone Laboratories not only for the courtesies extended at the time of the meetings, but also for the support and encouragement given the Acoustical Society in its formative period. It is believed that the encouragement of the Bell Telephone Laboratories has been of material assistance in promoting the gratifying success that has attended the organization of the Society.

PROGRAM

ACOUSTICAL SOCIETY OF AMERICA

Meeting New York City May 10th and 11th

May 9th joint session with Society of Motion Picture Engineers

All Meetings held in the Auditorium of Bell Telephone Laboratories,
463 West Street, New York City

MAY 9

8:15 P.M.

Joint session of the Society of Motion Picture Engineers and Acoustical Society of America.
Acoustic Facsimile (Demonstration Lecture), H. D. Arnold.

MAY 10

10:00 A.M.

Symposium on Methods of Measuring Absorption Coefficients.

1. The Measurement of Sound Absorption Coefficients by the Reverberation Method (30 minutes), Paul E. Sabine.

2. The Measurement of Coefficients of Sound Absorption by the Intensity Method and by the Oscillograph Method (30 minutes), V. O. Knudsen.

3. The Tube Method of Measuring Sound Absorption Coefficients (30 minutes), E. C. Wente.

Discussion (30 minutes), V. L. Chrisler, R. V. Parsons, Clifford M. Swan, Wallace Waterfall, and F. R. Watson.

Business Session (30 minutes).

12:30 P.M. to 2:00 P.M.—Luncheon.

2:00 P.M.

Nominations of Officers and Members of Executive Council.

1. Description and Demonstration of Artificial Larynx (20 minutes), R. R. Riesz.
2. Graphical Representation of the Reverberation Equation (20 minutes), F. R. Watson.
3. Variation of Sound Absorption Test Data (20 minutes), Wallace Waterfall.
4. The Hearing of Speech in Auditoriums (20 minutes), V. O. Knudsen.
5. Acoustical Properties of the Salt Lake Tabernacle (20 minutes), W. B. Hales.
6. An Instrumental Method for Measuring Reverberation (20 minutes), I. G. Ross.
7. An Instrumental Method of Reverberation Measurement, R. F. Noris and C. A. Andree.

8:15 P.M.

The Science of Musical Sounds (Demonstration Lecture), D. C. Miller.

MAY 11

9:30 A.M. to 12:30 P.M.

Symposium on Speech

1. Mechanism of Speech (20 minutes), G. Oscar Russell.
2. Theory of Vibration of the Larynx (20 minutes), R. L. Wegel.
3. Speech Power and Its Measurement (20 minutes), L. J. Sivian.

Discussion (15 minutes).

4. Frequency of Occurrence of Speech Sounds in Written English (20 minutes), Godfrey Dewey.

5. Frequency of Occurrence of Speech Sounds in Spoken English (20 minutes), N. R. French, Walter Koenig, Jr.

6. Effects of Distortion upon the Recognition of Speech Sounds (20 minutes), J. C. Steinberg.

Discussion (15 minutes), C. E. Seashore, W. H. Martin, and F. A. Saunders.

12:30 P.M. to 2:00 P.M.—Luncheon.

2:00 P.M.

Announcement of the Election of Officers.

1. A Spark Chronograph Developed for Measuring Loudness of Piano Tones (20 minutes), C. N. Hickman.

2. Scientific Methods of Piano Playing as Used in Making Ampico Records (30 minutes), Charles Fuller Stoddard, Research Laboratory, American Piano Co.

3. A New Just Scale with a Proof that an Additional Just Scale Is Impossible (20 minutes), John Redfield.

4. Human Factor in Piano Tone Production (20 minutes), W. B. White.

5. Methods and Apparatus for Measuring the Noise Audiogram (20 minutes), R. H. Galt.

6. Psychological Measurements of Annoyance as Related to Pitch and Loudness (20 minutes), Donald A. Laird.

Note: 165 members of the Society registered and were in attendance at this meeting. Others were present who did not register.

ABSTRACTS OF PAPERS

The Measurement of Sound Absorption Coefficients by the Reverberation Method. By Paul E. Sabine. Riverbank Laboratories. Geneva, Illinois.

The paper reports the results of an experimental study of the Reverberation Method. The assumptions made in the Reverberation Theory are critically examined. The fundamental equation

$$at_1 = 9.2\frac{V}{v}\log_{10}\frac{4E}{a_1^2 v}$$

affords two independent means of measuring the change produced in the quantity a for a given room, using a source whose acoustic power may be varied in known ratios. Outstanding differences in the experimental results of the two methods, are reported and possible explanations therefor are discussed. Effects of size and shape of test sample, and quality of test tones on measured values of the absorption coefficients are reported.

For detailed account of some of the matter reported reference may be made to Journal of the Franklin Institute for March 1929. Later work included experimental verification of a linear relation between the amplitude of vibration of the electrically driven piston source and the magnitude of the driving current.

The Measurement of Sound Absorption in a Room by the Intensity Method and by the Oscillograph Method. By Vern O. Knudsen, University of California at Los Angeles.

The differential equation for the flow of sound energy in a room of volume V yields two useful relations for determining a, the total sound-absorption in a room— one for the steady state, namely, $a=4E/vI_0$, and one for the decay, namely, $I=I_0e^{(av/4V)t}$. (E=rate of emission of sound-source, v=velocity of sound, I=average value of the instantaneous energy density, and I=the average value of the steady state energy density.) The advantages and limitations of different methods which utilize these fundamental relations are discussed. A method utilizing the steady state relation has been developed which exhibits considerable promise as a practical and precise means for measuring the sound-absorption in a room. The difficulties usually

encountered from variations of intensity owing to the stationary wave-pattern in the room are overcome by a variable frequency source which varies periodically between limits of 408 and 629 d.v. This shifts the interference pattern sufficiently to give approximately an average intensity of sound at any point in the room not too near the source. Measurements of a obtained by this method agree satisfactorily with values obtained by the reverberation method for a frequency of 512 d.v. Results obtained by Meyer and Just and also by the writer indicate that it is possible to obtain photographic records of the decay of sound in a room from which it is possible to calculate the total absorption in the room. Measurements of absorption obtained by this method agree, with a probable error of about ten per cent, with those obtained by the reverberation method

The Tube Method of Measuring Sound Absorption Coefficients. By E. C. Wente, Bell Telephone Laboratories, Inc.

The general principles underlying the tube method of measuring sound absorption can be derived conveniently from the analogous equations for the electrical transmission line. These equations show that the actual method of measurement is capable of many modifications, some of which have already been adopted by various experimenters. However, if reliable results are to be obtained, it is important that the apparatus be so designed that the propagation along the tube be rectilinear and the attenuation small, and that the tone be kept free from harmonics.

In the tube method the absorption is measured at perpendicular incidence, whereas in the reverberation method it is measured at random incidence. A theoretical study of the absorption of sound by porous materials as a function of the angle of incidence shows that in some cases there may be a considerable discrepancy between the values obtained by the two methods. The tube method may also give impracticable results for materials which are to be used in the form of large panels and absorb sound largely by virtue of inelastic bending rather than because of their porosity.

Description and Demonstration of an Artificial Larynx. By R. R. Riesz.

A description is given of the normal vocal tract and the action of the vocal cords in generating sound waves from the steady flow of air from the lungs. For persons whose larynx has been removed the flow of air

from the lungs is changed into sound by a device external to the body and this sound is converted into speech by introducing it into the mouth and carrying on the ordinary speech-producing motions with the lips, tongue, teeth, etc. A description is given of a recently developed vibrating reed type of artificial larynx for accomplishing the above. While using this instrument for producing a sustained vowel the rate of expulsion of air is about 125 cc per second and the excess pressure in the lungs is about .005 atmospheres, giving the rate at which the lungs do work as .06 watts. Using Saccia's measurements of the rate of radiation of acoustic energy from the lips of a speaker it is calculated that of the energy used by the lungs in forcing air through the glottis only about 0.17% is converted into useful acoustic energy.

Graphical Representation of the Reverberation Equation. By F. R. Watson. University of Illinois.

The growth and decay of sound in a room are usually expressed in terms of the *intensity* of sound, whereas it is more instructive to give these effects in *loudness* units, as perceived by the ear. Diagrams are drawn showing that the intensity of sound is greatly diminished by introducing absorbing materials in a room, but that the loudness is scarcely reduced at all. Other diagrams show the overlap of loudness of words in a speech. Further diagrams show the reduction of reverberation in a room when absorbing materials are introduced, and give information that is instructive in the experimental determination of coefficients of absorption.

Variation of Sound Absorption Test Data. By Wallace Waterfall, The Celotex Company, Chicago.

In the acoustical treatment of rooms, an accurate knowledge of sound absorption coefficients is necessary. Test data on Type B Acousti-Celotex from seven different laboratories is presented showing a variation from .40 to .47 in values obtained by the reverberation method and from .23 to .40 by the "tube" method. The values .23, .40 and .47 were all obtained on the same samples shipped to the different laboratories. The desirability of standardizing test methods is apparent.

To determine which absorption values should be used in engineering work, a study was made of a large number of completed installations. A small 512 cycle organ pipe was calibrated for acoustic output both at the University of Illinois and at the Bureau of Standards. The 3A Audiometer was used to determine the amount of noise, if any, in the

rooms and it was found possible to correct the measurements for the threshold shift caused by the noise. From the data, 45 installations are selected as typical. Using .47 as the coefficient, the total absorption in each room was calculated. In 24 of the rooms the test absorption was greater than the calculated absorption by an average of 9.7%. In 21 of the rooms the test absorption was less than the calculated absorption by an average of 12.1%.

The Hearing of Speech in Auditoriums. Vern O. Knudsen, University of California at Los Angeles.

A summary is given of the quantitative effects of loudness, reverberation, noise and the shape of an auditorium upon the hearing of speech in auditoriums. The average loudness of speakers is found to be about 50.7 db in small auditoriums (27,200 cu. ft.) and 45.7 db in large auditoriums (240,000 cu. ft.). The loudness of spech in large auditoriums is therefore at a critically low level, so that slight disturbances from reverberation or noise suffice to make hearing conditions unsatisfactory. The percentage articulation in an auditorium is ascertained by means of the formula,

$$\text{Percentage Articulation} = 96 k_l k_r k_n k_s,$$

where k_l, k_r, k_n and k_s are the reduction factors owing to diminution of loudness, reverberation, noise, and shape of auditorium, respectively. Methods and data are given for determining these reduction factors. A study of the combined effects of loudness and reverberation yields the optimal times of reverberation for the hearing of speech in auditoriums of different sizes. Thus, the optimal time for a room of 25,000 cu. ft. is found to be .80 second. The optimal time increases to .90 second for an auditorium of 100,000 cu. ft., and to 1.00 second for an auditorium of 400,000 cu. ft. It is shown that artificial amplification of speech is imperatively needed in auditoriums larger than 400,000 cu. ft., and is desirable in auditoriums larger than 100,000 cu. ft. A series of curves indicates the probable percentage of speech-articulation in auditoriums varying in size from 25,000 cu. ft. to 1,600,000 cu. ft. and having times of reverberation between .5 and 8.0 seconds. The results of the investigation make it possible to assign a quantitative rating to the acoustic quality (for the hearing of speech) of any auditorium, or to determine the physical requirements of an auditorium which is to attain a specified degree of excellence.

Acoustical Properties of the Salt Lake Tabernacle. By Wayne B. Hales. Weber College, Ogden, Utah.

Prof. Wallace Sabine's method of determining reverberation periods was used. Organ pipes in the tabernacle organ served as the source of sound. By opening and closing doors and windows the absorbing power of the building was found to be 630 open window units. The period of reverberation increased from 1.63 seconds for a note of 16 v.p.s. to a maximum of 5.21 seconds for 348 v.p.s. thence decreasing to 0.97 seconds for 8368 v.p.s. The constant of decay of sound, the mean free path, the paramoter k, and the volume of the building were determined by observing the variations produced in the reverberation by variations in the intensity of the sound.

The intensity of the sound was determined throughout the building and expressed in terms of the intensity meter from the source. It was found to vary from 100 per cent at near stations to 4.2 per cent at remote distances.

Articulation tests showed the percentage of syllables accurately received varied from 39 per cent for normal conversation voice to 59 per cent for preaching voice, to 80 per cent for electrically amplified voice. When the reverberation was reduced from 5 seconds for the empty building to 2.3 seconds when 3100 people were present good hearing increased from 52 per cent to 59 per cent for preaching voice in spite of presence of additional noises caused by audience.

An Instrumental Measurement of Reverberation. By Ira G. Ross. University of Illinois.

In measuring the time of reverberation in a room instrumentally, the usual practice of timing the decay of intensity from the maximum produceable by a given source to the threshold of audibility, possesses the disadvantage of requiring very sensitive sound detectors even when the "threshold" value is fixed at a relatively high level. It is proposed to measure, rather, the time of building-up of sound in the room from quiet conditions to some fixed fraction of the maximum intensity. This method presents the advantage of definitely fixing the low-intensity end of the time interval as the instant of starting emission at the sound source, while the end point lies in the region of high intensity and may be more readily determined instrumentally.

The sound intensity increase is followed by a condenser microphone, amplifier and valve voltmeter, a short-period galvanometer relay in the plate circuit of the latter being actuated when the sound intensity

reaches a predetermined level. The time of building-up, between the starting of the sound source and the closing of the relay, is equally as useful as the time of reverberation and from it the latter may readily be computed if desired.

An Instrumental Method of Reverberation Measurement. By R. F. Norris and C. A. Andree, C. F. Burgess Laboratories, Inc.

The usual method of measuring the period of reverberation with organ pipe and stop watch is uncertain, due to extraneous noises, variation in sensitivity of the ear, and other difficulties. Attempts to determine the period through measurements of the sound intensity level established by a calibrated source were unsuccessful.

A new method is here described in which a regular succession of equally timed and constant strength sounds are emitted. At desired instants in the intervals between the emission of sound the intensity is measured. From this data the period of reverberation is determined as $t_p = 6t'/\log_{10}(E_{ave}/E'_{ave})$, where t' is the time in which the intensity decays from E_{ave} to E'_{ave}, these three quantities being directly observable.

Chief of the advantages listed is the elimination of errors due to variations in the observer's acuity of hearing. The method is also applicable to the measurement of the sound-absorption coefficients of materials.

Mechanism of Speech. By G. Oscar Russell, Ohio State University.

Past experiments have been confined mostly to an examination of the recorded speech curves or of speech phenomena after the sound emits from the vocal passages. And the physiological cause was imagined. Various conflicting vowel theories have resulted.

A new method of attack has been carried out in the present investigation. Wheatstone *harmonic* or Willis *cavity tone* (erroneously called Helmholtz) theories are checked upon by direct physiological processes. Not only the forms but the exact dimensions of the speech cavities have been ascertained by means of X-rays, palatograms and larnygoperiskopik measurements, made simultaneously and instantaneously so that exact measurements of the vocal tract resonators are had in three planes making it possible to reconstruct these cavities with a high degree of accuracy. By attaching an artificial glottis to the model it should be theoretically possible to reproduce the vowel if the total air volume functioning as a resonator were responsible for vowel quality.

I am sorry to have to admit however that although 116 different types of reeds and other sound stimuli were used for such a purpose, the results were negative. All vowel cavities gave sounds which were much alike and but poor imitations of the original, just as are all other mechanical imitations of human speech which we have heard.

Theory of Vibration of the Larynx. By R. L. Wegel. Bell Telephone Laboratories, Inc.

A theoretical derivation of the equations of motions of the larynx is given and the analogy with the simple vacuum tube oscillator pointed out. This simple circuit consists of two meshes, the grid and plate circuits which are coupled by a unilateral mutual impedance, the amplifier constant, and a bilateral "feed back." Sustained oscillation takes place when these constants, which may have a variety of forms, have certain relative values. The vocal cords of the larynx are analogous to the grid, the glottis (the space between the vocal cords) to the plate circuit. A charge on the grid alters the resistance of the plate circuit and the displacement of the vocal cords alters the resistance of the glottis. The "B" battery corresponds to the lung pressure and the plate current to the air stream. The "feed back" of the larynx is the impedance of the lungs and that of the upper air chambers added to a "gyrostatic" coupling due to modulation of the mass of air in the glottis. The latter is most important and has no analog in electrical circuits.

Theory is given showing effects of relative values of the evaluated constants on sustained oscillation as in singing, and on subsidence or accretion of vibration as in speaking.

Speech Power and Its Measurement. By L. J. Sivian. Bell Telephone Laboratories, Inc.

The paper is chiefly concerned with the important speech power quantities—frequency spectra, distributions of instantaneous, average, syllabic and peak amplitudes, etc.—as they obtain in actual speech for a large range of voices, talking levels, and subject matters. The analysis is not nearly so complete nor so fine-grained as that which, in principle, can be derived from oscillographic records of individual speech sounds. Its advantage is in the speed with which data can be secured, under widely varying conditions and on a scale which warrants statistical conclusions. Some of the methods in use for measurements of this type are described. These include forms of apparatus developed prmarily for use in telephone plant work, as well as those developed for detailed

laboratory investigations. The data discussed are mainly those obtained in the laboratory with an integrator and with a peak meter. The former gives the average value of the instantaneous sound amplitudes; the latter gives the magnitude and frequency of occurrence of the peak values. The distribution of average and peak amplitudes in the frequency spectrum is obtained in addition to their values in the entire sound wave. Illustrative results are given for samples of speech, music and noise.

Frequency of Occurrence of Speech Sounds in Written English. By Godfrey Dewey.

There has been much qualitative research by fonetists and acoustic engineers dealing with the speech sounds of English, but until lately very little quantitative analysis, which is equally important in determining the relative importance of the various speech elements—words, syllables, and sounds—as measured by their relative frequency of occurrence.

The author's recent study entitled "Relative Frequency of English Speech Sounds" (Harv. Univ. Press) gives a classified quantitative analysis of the commoner words and syllables and all the sounds of 100,000 words of representative written English.

The word count data show that 10 words—*the of and to a in that it is I*—make up over 25% of all writing; that 100 words, including of course these 10, make up over 50%; and justify the conclusion that the 1,000 commonest words of English will make up about 75% of all ordinary non-technical writing.

The data on syllables and sounds, impossible to summarize briefly, are of far more important practical significance to the acoustic engineer than the word count data, and, unlike the latter, are at present the only authentic and comprehensive data in that field.

The Frequency of Occurrence of Speech Sounds in Spoken English. Norman R. French and Walter Koenig, Jr. American Tel. & Tel. Co.

This paper presents some results of a study of the words used in telephone conversations. These show that many words have a much higher relative occurrence in conversation than in written matter, and that conversation in general is characterized by a more intensive use of its commonest words than written matter. Curves are presented showing the number of different words found among a given total number of words, not only in individual conversations but also in samples

as high as 10,000 total words. Finally the paper shows the distribution of the fundamental sounds of speech in conversation and shows the differences in the relative occurrence of the speech sounds in conversation as compared with written matter, in nouns as compared with verbs, adjectives, etc., and in the few frequently used words as compared with the many infrequently used words. The results are based on about 80,000 recorded words.

Effects of Distortion Upon the Recognition of Speech Sounds. By John C. Steinberg. Bell Telephone Laboratories, Inc.

Physical studies of speech waves in conjunction with articulation tests have shown that in normal speech, the fundamental sounds are distributed in intensity and frequency. For example, the vowels are some 20 to 30 db greater in intensity than the fricative consonants, and require the range of frequencies, from a recognition standpoint, from 500 to 2500 cycles as compared with the range from 1500 to 8000 cycles for the fricatives. Some sounds, such as the vowels, tend to approach a "steady state" condition; others, such as the stop consonants, are of relatively short duration and almost "transient" in character. In both cases, however, recognition appears to depend considerably upon the initial period of growth and the final period of decay.

With these facts in mind, the primary effects of distortion may be readily understood. Distortion which tends to increase the intensity spread between sounds, or which effectively lowers the received speech level, decreases the articulation of the less intense sounds. Distortion which suppresses a portion of the frequency range, decreases the articulation of the sounds that are characteristic of that range. Systems which overload for the intense sounds tend to decrease the intensity spread, and may therefore increase the articulation of the faint sounds while decreasing the articulation of the intense sounds. Reverberation affects the various sounds differently, the greatest impairment occurring for the nasal and stop consonants. In no case did an increase in reverberation time cause an increase in articulation.

A Spark Chronograph Developed for Measuring Loudness of Piano Tones. By C. N. Hickman, Research Laboratory, American Piano Company.

This chronograph was developed for use in recording the loudness of blows from piano hammers as an artist is making an Ampico record.

It records accurately the velocity of each hammer at the moment it strikes the strings.

The chronograph, which is of the spark type, records on either stationary or slowly moving paper. A cylindrical drum with a spiral conductor distributes the time interval sparks at right angles to the motion of the paper.

A new method of making visible the spot, where the secondary spark punctures the paper, is described. This method permits the use of a very feeble current. A secondary spark energy sufficient to puncture the paper is all that is needed. The punctured spot may be made visible regardless of its size.

A simplified electrical circuit for producing the secondary spark is described.

Curves and data are given showing the relationship between the velocity of the hammer and the loudness of the piano tone in sesation units.

The chronograph with its auxiliary system of recording feeble sparks is applicable to many fields of scientific investigation.

Scientific Measurements of Piano Playing as Used in Making Ampico Records. By Charles Fuller Stoddard, Research Laboratory, American Piano Company.

This paper, illustrated by 30 lantern slides, was in the nature of a demonstration of the making of an Ampico record showing the various processes through which a record passes in its development between the time of playing to the finished reproduction.

The complete recording apparatus from the Ampico Laboratories was moved to the hall, and at the beginning of the reading of the paper, a record was taken of an artist's actual playing in person of Chaminade's Air de Ballet. Mr. Leslie Loth of New York was the artist. The paper explained in detail how every characteristic of the pianist's playing including quality of tone, feeling of "touch," exact loudness of every individual note, etc. was translated into note sheet performations which give a facsimile perfect enough to reproduce the emotional characteristics of the artist's performance.

After the reading of the paper, the Ampico reproduced Mr. Loth's playing of the Air de Ballet from a record which had been taken in the Ampico Laboratories three weeks previously and produced by the method explained in the paper.

Human Factor in Piano Tone Production. By William Braid White, Acoustic Engineer, American Steel & Wire Company.

A special form of portable Westinghouse oscillograph, using a Western Electric double-carbon-button 387-W microphone, and one stage of radio amplification, was used to obtain a large number of sound-wave photographs representing the sounds produced at the piano by various eminent artists, particularly Rudolf Ganz, John Powell, Ernest Schelling, and Olga Samaroff. The object was to discover how variations in attack upon the piano keys, transmitted to the strings through the hammers, affect (a) the intensity, (b) the quality of the sounds produced. The following conclusions were arrived at as a result of many experiments.

(1) The number of possible variations in key speed and in the control thereof is indefinitely great, much greater than is commonly appreciated by ordinary hearers, or by ordinary players, of the piano.

(2) An indefinitely great number of variations in the velocity of the hammer during its travel to the string is likewise possible.

(3) Every velocity value of the hammer, in a manner dependent among other things upon the physical make-up of hammer and string, produces a corresponding intensity value, and a parallel color value, in the sound.

(4) Faint sounds on the piano tend to converge in color value until at the minimum of intensity all sounds from bass to treble have the same sine wave form, that is, are perfectly simple sounds. Contrariwise, as intensity increases, the number of audible harmonics increases accordingly.

(5) It may therefore be concluded that every change in loudness involves a change in color, and every change in color requires a change in loudness; that is to say, tone color control on the piano is dependent upon control of intensity, that is, upon control of hammer velocity, and that again, on control of the speed and acceleration of the key.

The paper is illustrated with numerous sound-wave photographs, the evidence of which confirms the conclusions arrived at. The interpretation of these photographs is guided by the general rule that amplitude measures loudness, and shape of vibration-pattern measures quantity.

Methods and Apparatus for Measuring the Noise Audiogram. By Rogers H. Galt, Bell Telephone Laboratories.

The audiogram of a noise is the plot of the deafening effect of the noise versus the frequency of the sound which it is desired to hear. A portable apparatus has been developed for analyzing and measuring the deafening due to either acoustical or electrical noise by threshold observations upon a series of complex testing waves generated by a phonograph. Examples of noise measurements are given, and comparison is made between the fields of usefulness of the apparatus and of other noise measuring devices involving threshold observations.

Psychological Measurements of Annoyance as Related to Pitch and Loudness. By Donald A. Laird and Kenneth Coye. Colgate Univ. Phycological Laboratory.

The subjective experience of annoyance was measured with trained observers, using the tones of the 2-A audiometer as stimuli. A method of paired comparisons was first used which yielded results later verified by a method of equal annoyance gradients. The raw records in decibels indicate that annoyance increases as the pitch of the stimulus varies above or below the range characteristic of human speech. When the audiometer readings are related to Fletcher's contours of equal loudness the records indicate that for the lower half of the loudness series annoyance is directly related to loudness, but that for the upper half of the loudness series annoyance is greatly increased as the frequency of the stimulus is increased.

PROCEEDINGS OF THE BUSINESS MEETING

FRIDAY MORNING—MAY 10th, 1929

DR. HARVEY FLETCHER presiding.

CHAIRMAN: I think that our time for the business meeting has arrived. As announced, a temporary organization was set up, and officers and a Committee appointed to do two things: —one to provide a program, and a place of meeting for the Society, and one to draft up an organization, with a Constitution and By-Laws. I think we should hear first, a brief historical sketch, which Mr. Waterfall has prepared, telling you how the organization started. I think that he was one of the prime movers in getting the various people together, who organized this Society. I will ask him to read that now. (applause)

MR. WALLACE WATERFALL: I hope you will pardon the personal references in here. I have been asked a number of times just what was the incident which lead to the formation of the Society, and I have tried to cover it very briefly in this history, so you may know what has lead up to the present development.

I have another comment to make too: At the same time that the group of men gathered to form the organization, as it now is, there was considerable comment about the country regarding the possibility of forming an Acoustical Society, and the very large response to the first letters that were sent out is undoubtedly due to the fact that many men were thinking along these same lines. This helped greatly to bring the organization together, as rapidly as it has been brought together.

I have written this little historical sketch, and so I will read it as it is.

(Here take in the Historical sketch as read by Mr. Waterfall on the formation of the Acoustical Society.) (applause)

CHAIRMAN: I think that a report from the Committee on the Program is not necessary. The program speaks for itself.

The Committee appointed to draft a Constitution and By-Laws met, and as a result, the Constitution, a copy of which you have received was drafted. Permit me to say now that there have been some who have expressed a desire to become members to-day. Since we still have some copies of the Constitution, I am going to ask Professor Watson and Professor Knudsen to pass them out to those who have not received a copy, so that you can now become members if you so desire and take part in this meeting, as Charter members. After this meeting, the Society will be considered to be formally organized.

We are also passing around some pads of paper, and we want you to write your names and addresses on them, to check up with our list of members.

While that is being passed around, I am going to ask the Secretary to read for your approval, the Constitution. Inasmuch as this Constitution has been signed by 376, now official members, and by the show of hands we have had, we will, I think, have over 400, before this meeting is adjourned—they have signified their willingness to abide by this Constitution so I don't think it needs any consideration, except the reading of it. So we will ask Mr. Waterfall to read the Constitution. Mr. Waterfall (reading)

CONSTITUTION of the ACOUSTICAL SOCIETY OF AMERICA

WE, THE UNDERSIGNED, desiring to form a society or association for the purpose hereinafter named, do hereby subscribe this constitution for that purpose as follows:

ARTICLE I—NAME

The name of the society shall be the "Acoustical Society of America"

ARTICLE II—PURPOSE

The purpose of the Society shall be to increase and diffuse the knowledge of acoustics and promote its practical applications.

ARTICLE III—MEMBERSHIP

The society shall be composed of the subscribers to this constitution at the date of organization and such other persons or corporations as shall be hereafter admitted to membership in accordance with the provisions of the by-laws.

The membership shall be classified as follows, honorary members, sustaining members, and members, as provided in the by-laws. Sustaining members may be either persons or corporations; all other members must be persons. The voting power shall be restricted to and vested exclusively in the members constituting the fellows of the society.

ARTICLE IV—EXECUTIVE COUNCIL

The affairs of the society shall be in charge of an executive council which shall consist of the president, vice-president, secretary and treasurer and six fellows to be elected as provided in the by-laws.

ARTICLE V—OFFICERS

The officers of the society shall consist of a president, vice-president, a secretary, and a treasurer, who shall be chosen in the manner provided in the by-laws.

ARTICLE VI—AMENDMENTS

This constitution may be altered or amended at any meeting of the society when a proposed amendment, as submitted by the executive council, is favored by two-thirds of all of the fellows present thereat voting upon such amendment. A copy of every proposed amendment shall be mailed to each fellow of the society at least thirty days before a vote is to be taken thereon. All voting shall be by ballot.

This constitution may be subscribed to in any number of counterparts, all of which shall constitute one and the same instrument, irrespective of the time or times when the same, or any of the counterparts, was or may be subscribed to.

IN WITNESS WHEREOF, the undersigned have subscribed to this constitution as of the ———— day of ————, 1929.

(Signatures)

CHAIRMAN: This was drawn up in this legal language, because we felt that we wanted to pave the way for organizing into a corporation. We are going to have financial support, I am sure, from a great many of the corporations, who are interested in acoustics, and for that reason we felt it necessary to incorporate the Society. This accounts for the legal language used in the Constitution.

Now, I have been asked several times about dues, and people have been trying to give us money for membership dues. We haven't accepted it. We felt that the Society had not been formed, and we had not right to accept money. But before the day is over, I hope we will be in a position to accept dues, and our treasurer will accept it, and we will make an announcement to that effect in due time.

I think, however, that we ought now to consider the by-laws, and before I ask Mr. Waterfall to read the By-laws, I should like to make an apology for a mistake which was made in the printed copy. The printed copy does not represent the Committee's view on how the by-laws should read. In one place, which will be pointed out, it reads "only the fellows will receive the publications, etc." It was the sense of those who drew up the by-laws that all the publications, and all the meetings should be open to all the members. We will make that correction in the reading of it so it will be slightly different than the copy which you have.

The by-laws are presented for your discussion, correction, and adoption. They will not be official until you adopt them. You can make any changes you wish. We will have them read, article by article, so we will have the time to think of part and change it if you so desire.

MR. WATERFALL: (Reading and pausing between each article to permit discussion)

BY-LAWS

ARTICLE I. NAME AND PURPOSE OF THE SOCIETY

The name and purpose of this society shall be those stated in the constitution.

ARTICLE II. MEMBERSHIP

Section 1. Eligibility. There shall be four classes of membership, viz.: honorary, fellows sustaining members, members.

Any person or corporation interested in acoustics shall be eligible for membership in this society. Every person or corporation desiring to become a member of this society shall in person or by authorized representative sign an application for such membership.

Any person of outstanding eminence in theoretical or applied acoustics may be elected to honorary membership.

Any person who shall have rendered conspicuous service and made notable contributions in acoustics shall be eligible to fellowship in this society.

Any person or corporation contributing Fifty Dollars ($50) or more annually may be elected a sustaining member.

Section 2. Election. Applications for membership must be signed by two or more fellows or members and forwarded to the secretary. Applicants shall be admitted to membership when their applications have been approved by three or more members of the executive council.

All classes of membership other than honorary shall be elected by a majority vote at a regular or special meeting of the executive council. Honorary members shall be elected only by a unanimous vote of the executive council.

Section 3. Privileges. The right to vote at elections, and hold office shall be restricted to and exclusively vested in the membership constituting fellows. All other classes of membership, other than honorary, shall have the same privileges as fellows, except as herein otherwise provided.

Honorary members shall have such privileges as may be granted by the executive council.

Section 4. Termination of Membership. Membership in the society may be terminated at any time by a majority vote of the executive council. Failure to pay dues for one year would be deemed sufficient cause for termination of membership.

ARTICLE III. DUES

The annual dues of the society shall be Seven Dollars and Fifty Cents ($7.50) for fellows and Five Dollars ($5.00) for members.

Disbursements shall be made by the treasurer when such disbursements are approved by a two-thirds vote of the executive council.

MR. STODDARD: I would like to see the dues made the same for members and fellows. I do not see why it should cost $2.50 to be a fellow. The only difference is the vote, and it is not ethical to pay for a vote. (laughter)

CHAIRMAN: Do you make a motion to that effect?

MR. STODDARD: I make a motion that $7.50 be the dues for all members, excepting the honorary members.

MR. MUNROE: I second the motion.

A VOICE: I presume that Mr. Stoddard also excepts sustaining members.

MR. STODDARD: Sustaining members are not excepted. They are expected. They pay $50.00 or over. That seems to cover it. So there is no need of an exception. We are going to

have a Journal. It is going to be very expensive to put that out. We think it will probably cost about $7.50, or perhaps more for each subscription of the Journal, and we are going to try to meet that expense from the donations of the sustaining members, which will be the corporations, whom a great many of the gentlemen represent, and it does'nt seem too much to pay $7.50 from my standpoint.

MR. ANDERSON: (Toronto)—There is no provision made for the initiation fee. I think that it is customary in associations of this kind to have a preliminary initiation fee for those coming in. I don't want to boost the dues up, but I call attention to the fact that that is omitted.

CHAIRMAN: That was omitted consciously, and discussed by the Committee. Of course this Committee spent some time on this. These questions were discussed.

MR. ANDERSON: One other point: Mr. Waterfall said that failure to pay the dues for one year entitled the member to be expelled.

CHAIRMAN: It didn't say that he would be expelled, but it would constitute a sufficient cause to have him expelled. He could only be expelled by a vote.

MR. ANDERSON: Isn't that drastic?

CHAIRMAN: Well, that was also discussed. That is all I can say about it. Are there any other remarks on this?

A VOICE: I like pretty nearly everything about Mr. Stoddard, except this motion. I am sorry I have to oppose it. I have a great deal of sympathy for the poor acoustic students in the laboratories, to whom an extra $2.50 means quite a good deal.

MR. MILLER: I should like to second the last remark. I am in favor of having a large membership. That is what I think we want.

CHAIRMAN: We have received communications, unsolicited from quite a number of corporations asking that they might be permitted to give us some money towards this Society. That is rather an unusual condition, but it has actually happened.

MR. HEWLETT: I think that if this comes to a vote, I will vote "No." I don't know of any one of a great number of Scientific and Engineering Societies that does not make a differential for fellows, which gives them the privilege of running the outfit, and I think you open up quite a large question—quite a lot of psychology—whether a lot of these pople you want to come in, and you wish to have, will pay as much if deprived of the vote. Quite true, you don't pay for a vote. You might say that it is an added burden put on these members for running it. You ought to give them a salary for running it. I think it is a matter of general experience with other similar organizations, over many years. I think we should not experiment or change what the executive committee have set up.

CHAIRMAN: Is there any further discussion?

A VOICE: Question.

CHAIRMAN: It is a question of changing the by-laws as read, so that the dues will be $7.50 for members, and $7.50 for fellows. All in favor, make it manifest by saying "Aye." Opposed: "No." (lost) It will remain as recommended by the Committee. The "No's" are very strong.

MR. WATERFALL: (Continuing the reading of the By-Laws)

ARTICLE IV. OFFICERS

The officers of the society shall consist of a president, a vice president, a secretary and a treasurer.

Section 1. Eligibility. Any fellow is eligible for election to any office in the society.

All officers except members of the council shall be elected for a period of one year.

Two members of the council shall be elected each year and shall hold office for a period of three years.*

All officers shall take office two weeks after election.

After serving two consecutive terms the president shall be ineligible for re-election. After serving one term members of the executive council shall be ineligible for immediate re-election.

Section 2. Election. Election shall be by ballot. Nominations shall be made by a nominating committee appointed by the president and approved by the council. This committee shall consist of five fellows of the society, at least one of whom shall, when present at the meeting, bea past president of the society. The name of any candidate suggested by more than ten fellows shall appear on the official ballot along with those nominated by the nominating committee.

ARTICLE V. MEETINGS

The meetings of the society and the executive council shall be held at such times and place as may be designated from time to time by the executive council.

ARTICLE VI. ORDER OF BUSINESS

The order of business shall be determined by the executive council.

ARTICLE VII. EXECUTIVE COUNCIL

The affairs of the society shall be managed by an executive council which shall consist of the president, vice-president, secretary and treasurer and six members to be elected as provided in these by-laws. Each member of the council shall be a fellow in good standing at the time of election.

Any vacany occurring in the executive council may be filled for the unexpired term by a majority vote of the remaining members of the executive council.

ARTICLE VIII. PRESIDENT

The president shall, if present, preside over all meetings of the society and the executive council. In case of the absence of the president, the vice president shall have all the powers and perform all the duties of the president.

ARTICLE IX. SECRETARY

The secretary shall attend all meeting of the society and the executive council and shall keep a true and faithful record thereof.

DR. SABINE: Is there any place in the Constitution which provides for the travelling expenses of officers? It occurred to me in connection with the Secretary.

CHAIRMAN: There has been no provision up to the present time.

DR. SABINE: Is it ordinarily considered the proper thing to do to include that in the Constitution?

CHAIRMAN: At the present time, that will rest in the hands of the Executive Council. But it is expressly stated that no officer of the Society shall receive remuneration for his services.

MR. WATERFALL: (Concluding the reading of the By-Laws)

ARTICLE X. TREASURER

The treasurer shall have charge of and receive all the funds and securities of the society. He shall deposit the funds to the credit of the society in such a bank or trust company as the executive council shall direct and he shall disburse the same only upon the approval of the executive council.

ARTICLE XI. AMENDMENT

These by-laws may be altered or amended at any meeting of the society. They may also be altered or amended at any meeting of the executive council by a majority vote of the whole number of members thereof, provided that in the call for such meeting, notice shall have been given to each member of the executive council that a proposition to amend the by-laws would be submitted at such meeting, which notice shall set forth the substance of the proposed amendment.

* At the first election two members of the council shall be elected for a term of one year, two for a term of two years, and two for a term of three years.

A VOICE: There seems to be a slight inconsistency. It calls for two-thirds of a vote of the Council on all financial transactions. Is that correct?

CHAIRMAN: I think so.

A VOICE: Then it can be amended by a majority vote?

MR. WATERFALL: No; it is not a two-thirds vote. (reading again)

A VOICE: The previous clause to that is what I had in mind.

MR. WATERFALL: There is no previous one that I can find.

CHAIRMAN: You have heard the by-laws as read, what is your pleasure?

MR. MILLER: The last article don't suit me. I think the Society ought to determine the by-laws. The Executive Council should act under the authority of the Society. But the By-laws are a part of the organization of the Society. It is not a matter of business. I move that the amendments to the by-laws shall be made at a vote of the Society upon recommendation of amendment from the Executive Council.

A VOICE: I second the motion.

A VOICE: It should have at least a two-thirds vote.

CHAIRMAN: The Constitution?

A VOICE: I mean the by-laws too.

CHAIRMAN: I don't see why. The by-laws are just a means of doing business. A majority vote ought to suspend them.—

All in favor of this amendment to the by-laws, make it manifest by saying "Aye;" opposed, "No." (Carried)

CHAIRMAN: Are you ready to vote on the by-laws as they stand amended with this amendment?

MR. MILLER: I move that they be adopted as amended. (seconded, and carried)

CHAIRMAN: Now we have a Constitution and By-Laws. They provide for electing officers in a certain way, and we can't quite operate, because we haven't any fellows. We will have to go ahead and do so by suspending them, it seems to me. (laughter) I think we ought to take some measures now toward the election of a set of officers. I don't mean to elect them, but how shall we proceed to nominate them? We ought to go at this rather deliberately. There are the four officers, and six members of the Executive Council.

A VOICE: In the absence of a President, the presiding officer would function.

CHAIRMAN: If you have no objection, I will act according to the by-laws, and appoint a Nominating Committee to recommend nominations for these various offices. It will make a report at this afternoon's meeting. We will receive any other nominations from the floor, by any one who is not satisfied with the recommendations of the Committee. We will have an election at that time. If there is no objection to this we will proceed in that way. Have you any objections? I don't want to railroad this thing through, because it is very important. With approval then I will appoint the Nominating Committee. For Chairman I suggest Professor Dayton C. Miller. As Members of the Committee, Professor Anderson, (Toronto), Dr. Hewlett of the General Electric Co., Professor Saunders, Mr. Clifford Swan, Dr. Sabine, Professor Watson. I think you will have confidence in these men for selecting the officers of this Society.

Now, if there are any further suggestions, or if there is any further business, we want to attend to it before going to lunch. The appointment of this nominating committee does not prevent any one here who wants to get together in some sort of a rally, or what not, to get a new ticket which they think will be a better one than this Committee will select. I would like to ask if Mr. O'Rourke is here. If so, he will make an announcement about Luncheon.

(At this point Mr. O'Rourke made several announcements).

WHEREUPON, the first session was adjourned at 1:35 P.M. for lunch, to reconvene at 2:15 P.M. the same day and place.

(LUNCHEON)

REPORT OF NOMINATING COMMITTEE AND ELECTION OF OFFICERS

FRIDAY AFTERNOON—MAY 10TH, 1929

DR. HARVEY FLETCHER presiding.

CHAIRMAN: Now, I think it is only fair to those who are leaving, who expected the report of the nominating committee to come in early this afternoon, to hear this report now. Then we will go ahead with our regular meeting, and discuss this paper. If you have no objections we will proceed to hear this Committee's Report. Professor Miller was the Chairman of the Committee.

MR. SAUNDERS: Professor Miller invited me to take his place. It is a subject of extraordinary delicacy, and I beg your indulgence for various reasons.

The Nominating Committee begs to present the names of the Temporary officers for a continuation of their office. Feeling that Dr. Fletcher, Professor Knudsen, Mr. Waterfall, and Mr. Stoddard have shown themselves to be adequately suited for the positions they have temporarily occupied, we, for the Society, desire to see them continue. That requires no apology, and no particular delicacy of utterance, but the rest is different. I might say that some people of more or less intelligence thought a good deal about various questions involved in the nominating of the Executive Council. It was considered desirable to have people representing the different branches of industry in the Society,—the different educational and research institutions involved,—the proper geographic distribution. All those questions had been considered to a certain extent, I am afraid, even in the selection of the Nominating Committee. The Nominating Committee, with utmost modesty, seemed to be in entire agreement in regard to this selection, with the exception of myself, and that is the reason that I can speak. (laughter) We therefore nominate, having really very solemnly considered the question, and considered the aspect which we would present to the members, we nominate for the Council for a one year term, Dr. Paul E. Sabine, Dr. J. P. Maxfield; for a two year term, Dr. C. W. Hewlett and Professor G. R. Anderson; for a three year term, Dr. D. C. Miller, and Professor F. R. Watson. (applause)

CHAIRMAN: I didn't think these men would be very backward when I appointed them on this Committee. (laughter) I assure you that there has been no collusion in this. (laughter) This, however, does not close the matter. This is simply a means of expediting business in getting a ticket. Here are some people that have been nominated. You have heard them nominated. Now, we want to make this a very democratic organization, so we now throw the thing open to anyone who wishes to put any other names in nomination. I will say here, when the nominations are finished, that we will vote by a secret ballot, so there will be no embarrassment as to how you voted. I will now call for nominations from the floor. Don't be backward. As temporary President of this organization, I thought I had done my bit already.

A VOICE: I move that the nominations be closed.

A VOICE: I second the motion.

CHAIRMAN: It has been moved and seconded that the nominations be closed. I will put the question if there is no further discussion. All in favor, say "Aye;" contrary, "No." (Unanimous). So ordered.

Then the matter of voting is a mere formality. If you have no objection we will suspend the rules in the by-laws, which we adopted regarding voting, and put the thing to a vote here. Is there any objection to that procedure? Inasmuch as there is only one man nominated for

each office, I suppose we can do that. All who favor these temporary officers as permanent officers, please signify by saying "Aye." (Carried unanimously)

We take it then, that you are back of us as officers—all who are members. I assure you that this Society has already grown to quite a large membership—over 400. And it is represented by the men in America who stand for doing things in acoustics.

You may be interested to know that we have as members of this Society two Past Presidents of the American Institute of Electrical Engineers, Dr. Carty, and Dr. Jewett; one Past President of the American Physical Society, Professor Miller; one past president of the Optical Society of America, Professor Richtmeyer. There may be others. But at least you see that the men who have done things and stood for things in the scientific and engineering fields, are backing this organization. And I wish to say that not only the men of America which are doing great things along this line, but the corporations who are interested in acoustics all over America have expressed a desire to back this thing to the limit.

A rather strange thing has happened. We have received four or five letters from corporations asking us if they couldn't donate some money to help this Society along. We have refused money up to date, but now we will go after it. I will make an announcement on behalf of the Treasurer, Mr. Stoddard. Immediately after this meeting you will find him up here at a desk, ready and glad to receive your annual dues as members.

The question of Election of Fellows, will be taken up by the Executive Council.

There are some other business matters that must be attended to by the Executive Council and I think that if it is possible, I should like to meet the Council to-day. There are a large number of questions to be considered, one being the question of the publication of a Journal. I feel sure that we have sufficient financial support already so that we can promise you that the proceedings of this meeting will come out as the first publication of the Society. What the future plans will be, of course, will be put in the hands of a Committee.

The question of the next meeting of the Society should also be discussed tonight, so we can make announcements tomorrow.

(Announcements were made by the Chairman on various subjects.)

GRAPHICAL REPRESENTATION OF THE REVERBERATION EQUATION

By F. R. Watson
University of Illinois

The growth and decay of a steady sound in a room are usually expressed in terms of the intensity of sound, as pictured in Fig. 1. The intensity rises rapidly at first, then more slowly until it reaches a maximum, when as much energy per second is absorbed as is generated. It dies out in the reverse order,—rapidly at first, and then more slowly until it becomes inaudible.

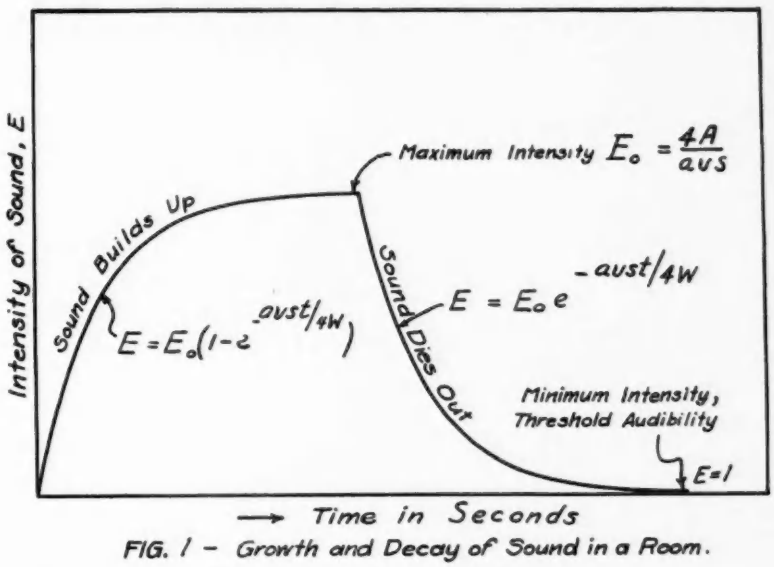

FIG. 1 – Growth and Decay of Sound in a Room.

Fig. 1. *Intensity-time curve showing the growth and decay of sound in a room.*

While such intensity-time curves give correct pictures of the intensity, they are not as instructive as if given in terms of loudness, which is proportional to the logarithm of the intensity. That is, since a person perceives the loudness of sound rather than the intensity, he will understand the action of sound better if loudness-time diagrams are used.

For example, consider the reverberation in a room before and after absorbing material is added, for which the intensity-time curves are

shown in Fig. 2. It is to be noted that the maximum intensity in the room after the material is added is much less than before, a fact that leads to the erroneous conclusions that the loudness is diminished by

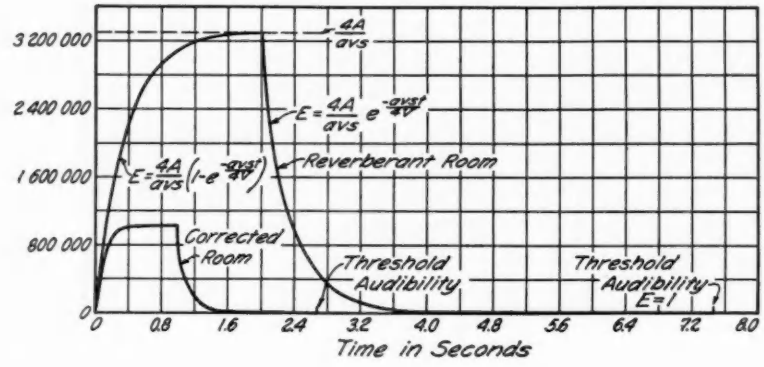

Fig. 2. *Intensity-time curves showing the intensity of sound in a room before and after sound-absorbing material is added.*

the absorbing material. The loudness-time plot in Fig. 3 gives a more correct picture, and shows that the loudness in the deadened room is scarcely less than in the reverberant room.

If now a connected discourse in a reverberant room is studied, we get some interesting new ideas. The intensity-time curves are shown in

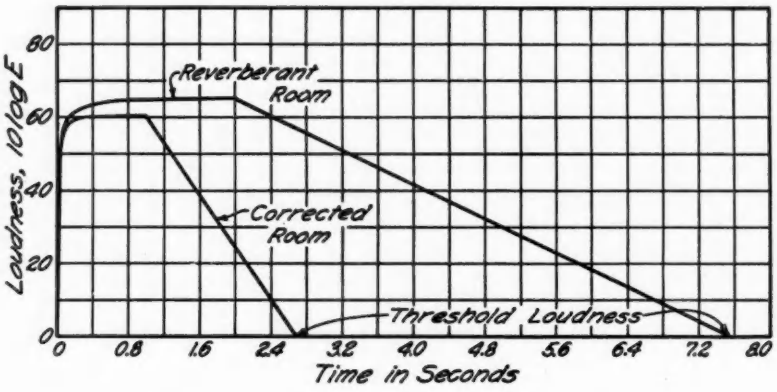

Fig. 3. *Loudness-time curves for intensity curves shown in Fig. 2.*

Fig. 4. The first curve at the left is a reproduction of the upper curve in Fig. 2. Assuming that the time taken to say each word is 0.3 second, with a pause of 0.1 second between words, we have the series of

lower curves (solid lines) in which the intensity rises for 0.3 second and then dies out rapidly. There appears to be but little overlap of successive sounds, whereas it is well known that it is difficult to understand a speaker in a reverberant room because his words overlap.

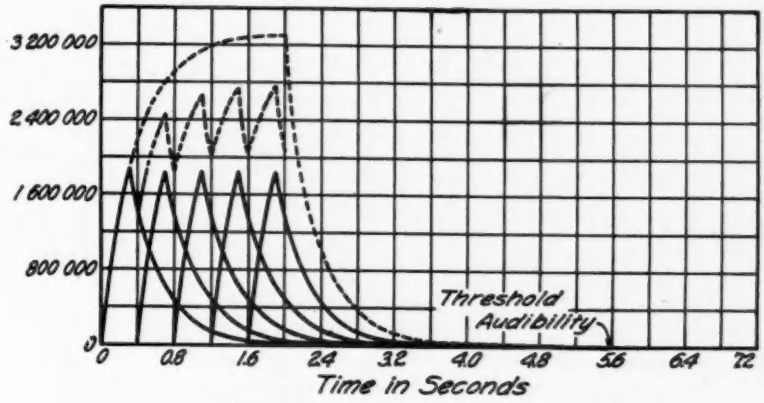

FIG. 4. *Intensity-time curves for successive words in a speech.*

Fig. 5 gives a loudness-time diagram for the sounds of Fig. 4, that shows clearly the overlapping of sounds, which are of nearly equal loudness at any instant.

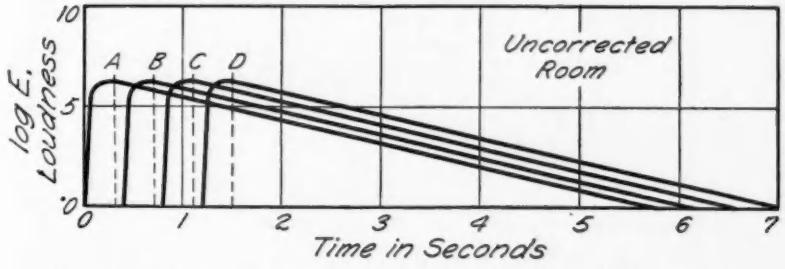

FIG. 5. *Loudness-time curves showing the overlap of words in a speech.*

Referring again to Fig. 4, the resultant intensity is given by the zigzag dotted line which is obtained by adding the intensities of the separate sounds at any instant. The corresponding resultant loudness is shown in Fig. 6, in which it is seen that the variation in loudness is only about 3 per cent, whereas in Fig. 4 the resultant intensity varies as much as 30 per cent.

If a reverberant room is deadened with absorbing material, the sounds die out quicker and the overlap of the sounds pictured in Fig. 5 is reduced as shown in Fig. 7.

An audience is efficient as an absorber of sound because of the clothing worn. Fig. 8 shows the effect of the audience.

The overlap of loudness of sounds in a speech furnishes an important means of studying the reverberation in rooms. The correction of de-

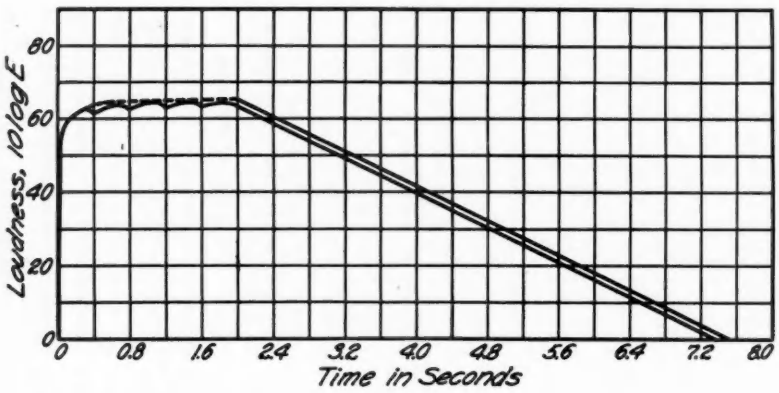

Fig. 6. *Loudness-time curves showing resultant loudness of words of a speech.*

fective acoustics in rooms is usually brought about by reducing the time of reverberation by the introduction of absorbing material and this, as we have just seen, reduces the overlap of successive sounds. If we can determine the average time between words in a speech and

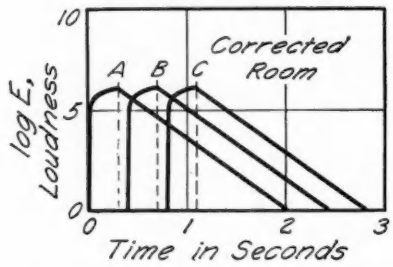

Fig. 7. *Loudness-time curves for sounds in a room deadened with absorbing material.*

also the amount of overlap of loudness that allows the separate words to be understood easily, we fix what may be called the critical time of reverberation. That is, the time of reverberation of any room should be reduced at least to this critical time, if words are to be easily understood. If the time is greater than the critical value, words will be blurred; if less than the critical value, the words will be more clearly sep-

arated. This critical time is involved in the general question of ideal auditorium acoustics,[1] for which additional information is desired.

For example, suppose that the average time interval between words is 0.1 second, and that the loudness of a word A (Fig. 7) reduces to one-half the loudness of word B as the letter reaches its maximum value, and suppose that the word B is easily understood under the cir-

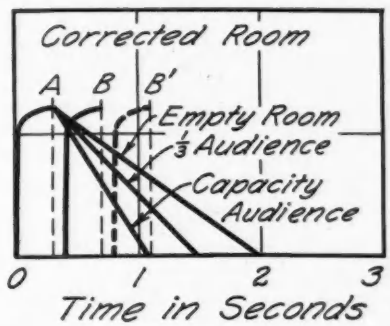

FIG. 8. *Reduction in time of reverberation by an audience.*

cumstances. Then the time taken for the word A to die out (seconds) will be the critical time of reverberation. Any time greater than this value in an auditorium would give defective acoustics, while a time less than seconds would give greater clearness of speech. These values are not determined with any great precision at the present time, but if they are found, the critical time can be determined and should apply to any auditorium, independent of size.

DETERMINATION OF SOUND-ABSORBING COEFFICIENTS

Turning now to the problem of the determination of absorbing coefficients by the reverberation method, we obtain some further instructive information. The experimental procedure is to fill a reverberant room with a pure, controlled sound, then to measure the time taken for it to become inaudible after the source of sound is stopped. This is done for the bare room and also with absorbing material present. The increase in absorption from a_0s_0 (bare room) to $a_0s_0+a_1s_1$ (with material installed) reduces the initial loudness from $\log 4A/a_0s_0v$ to $\log 4A/(a_0s_0+a_1s_1)v$, and the time of reverberation from t_0 to t_1. The loudness-time curves for these two cases are shown in Fig. 9. The equation used for

[1] Ideal Auditorium Acoustics, by F. R. Watson. Journal of Amer. Inst. of Architects, vol. 16, p. 259, 1929.

calculating the absorbing coefficient a_1 is as follows:[2] $a_1 = a_0 s_0 (t_0 - t_1')/s_1 t_1'$ where the time t_1 is increased to t_1' so that the initial loudness will be the same as for the room without absorbing material. Suppose now that a second observer has more sensitive hearing and can detect threshold sounds fainter than before, as given by the horizontal dotted line; will he get different coefficients than the first observer? The answer is "No." The coefficients depend on the ratio of the times $(t_0 - t_1')$ to t_1', and this ratio will be the same for any horizontal intercept, so that all observers should get the same coefficients, provided the time is taken for the bare room each time as well as for the material under test.

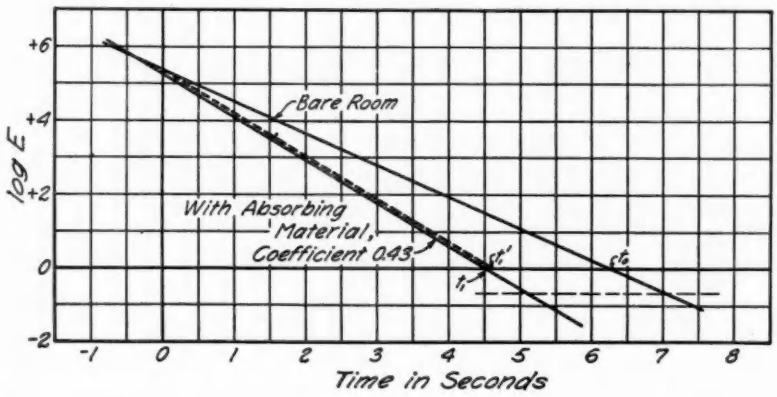

FIG. 9. *Loudness-time curves for bare room and with sound absorbing material added.*

Another necessary feature of the reverberation method in obtaining absorbing coefficients is the determination of the absorbing value of the bare room $(a_0 s_0)$. What is done to measure the time of reverberation for several different intensities (A) of the sound used, for which we have the reverberation equations

$$E = \frac{4A_0}{a_0 s_0 v} e^{-a_0 s_0 v t_0/4V} = \frac{4A_0'}{a_0 s_0 v} e^{-a_0 s_0 v t_0'/4V}$$

so that the absorption $a_0 s_0$ may be calculated from the relation:

$$a_0 s_0 = \frac{4V 2 \cdot 3}{v} \frac{(\log A_0 - \log A_0')}{(t_0 - t_0')}.$$

[2] The Absorption of Sound by Materials, Bulletin 172, Univ. of Ill. Eng. Expt. Sta.

The ratio $(\log A_0 - \log A_0')/(t_0 - t_0')$ is obtained from the loudness curve in Fig. 10, where the intensities A are given in terms of $\phi/\cos 2\phi$ as measured by a Rayleigh disc.[3] The loudness curves are shown in Fig. 10 for both the bare room and with absorbing material present.

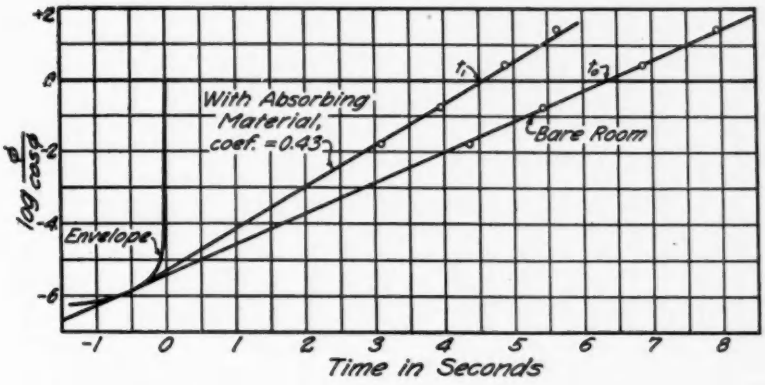

FIG. 10. *Loudness-time curves giving the relation between the loudness of the source of sound and thet ime of reverberation.*

It may be shown that these lines meet in a point for which t is negative; also that the lines are tangent to an envelope. The analysis is as follows.

$$E = \frac{4A}{avs} e^{-avst/4V} \tag{1}$$

let E_T = threshold value of E, where t = corresponding time.

$$E_T = \frac{4A}{avs} e^{-avst_T/4V} \tag{2}$$

$$\log E_T = \log A + \log \frac{4}{avs} - \frac{avst}{4V} \tag{3}$$

bare room

$$\log A = \log E_T - \log \frac{4}{a_0 v s_0} + \frac{(a_0 v s_0 t_0)}{4V} \tag{4}$$

bare room + material $(a_1 s_1)$

$$\log A = \log E_T - \log \frac{4}{(a_0 s_0 + a_1 s_1)v} + \frac{(a_0 s_0 + a_1 s_1) v t_1}{4V} \tag{5}$$

[3] loc. cit. "The absorption of sound by materials," page 16.

from 3, if log A is plotted against t, with E_T and as constant, we get a straight line. If now the absorption, as, is varied, we get family of straight lines, of which 4) and 5) are members. The lines 4) and 51) intersect (log E_T and log A constant) in t_p(subtracting 4 from 5)

$$0=\log\frac{4}{a_0s_0v}-\log\frac{4}{(a_0s_0+a_1s_1)v}+\frac{vt}{4V}(a_0s_0+a_1s_1-a_0s_0)$$

$$t_p=\left(\log\frac{a_0s_0}{a_0s_0+a_1s_1}\right)\frac{4V}{v}(a_0s_0+a_1s_1)-a_0s_0$$

$$=\frac{4V}{v}a_1s_1\log a_0s_0-\log(a_0s_0+a_1v_1)$$

(always negative, since $\log(a_0s_0+a_1s_1.) > \log a_0s_0.$)

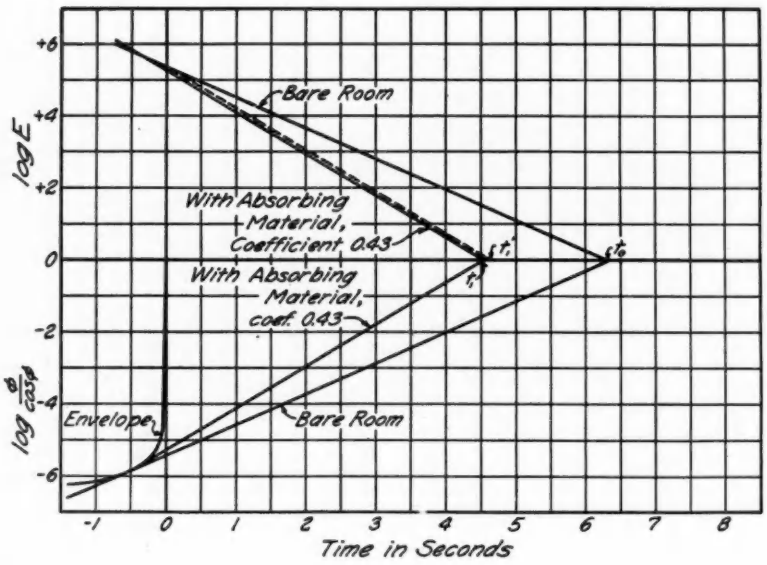

FIG. 11. *Loudness diagram symmetry between the reverberation in a room and the loudness of the source of sound.*

The envelope for the family of lines is:

$$\log A-\log E_T+\log\left(-\frac{t_p}{V}\right)+1=0 \quad \text{envelop}$$

since t_p is negative, $-t_p/V$ is + and $\log(-t/V)$ is real.

The slope of 4) is:

$$\tan\theta_1=\frac{d(\log A)}{dt}=0-0+\frac{a_0vs_0}{4V}$$

The slope of 5) is:

$$\tan\theta_2=\frac{d(\log A)}{dt}=0-0+\frac{(a_0s_0+a_1s_1)v}{4V}\,.$$

Angle θ between 4) and 5) is

$$\tan\theta=\frac{\tan\theta_1-\tan\theta_2}{1+\tan\theta_1\tan\theta_2}\,. \qquad (8)$$

Similar equations may be developed for the reverberation, as shown in Fig. 9, and prove to be the "mirror" of the loudness curves in Fig. 10; the combined curves being shown in Fig. 11.

Inspection of Fig. 10 shows as the absorption in the room increases, the loudness lines get steeper, until the limit is reached where the line is vertical corresponding to infinite absorption, for which the time of reverberation would be zero for any loudness of the source. On the other hand, if the absorption gets less, the loudness line gets more nearly horizontal and in the limit, $a_1s_1=\infty$, and $t_p=-\infty$, Equation 6. Similar deductions may be made for the lines in Fig. 9.

THE HEARING OF SPEECH IN AUDITORIUMS

By Vern O. Knudsen
University of California at Los Angeles.

Ever since the monumental work of W. C. Sabine on reverberation there has been a growing tendency, especially in America, to rate the acoustic quality of an auditorium almost solely in terms of its time of reverberation. It is true that reverberation (which determines the rate of growth and decay of sound in a room) has been, and yet is, the most important factor in determining the acoustic properties of a room. However, reverberation is not the only factor affecting the acoustic properties of an enclosure. Thus, the size and shape of the room, and the presence of extraneous noise, all contribute to the resulting acoustic quality. It would seem desirable therefore to evaluate the acoustic merit of a room in terms of all of these factors.

It is not a simple matter to give a quantitative rating to a room which is to be used for music, since so much depends upon the musical taste and dispositon of the listeners. It is, however, a relatively simple matter to give a quantative rating to a room which is to be used for speaking, since our primary concern is how well we hear the spoken words of the speaker. The most feasible scheme for such a rating is probably the one used by telephone engineers for testing speech-transmission over telephone equipment, which goes by the name of "articulation tests." The "percentage articulation' of a telephone circuit sigifies what percentage of typical speech-sounds can be heard correctly when transmitted over the circuit. Thus, if a speaker calls out 1000 meaningless speech-sounds into the transmitting end of the circuit and an observer at the receiving end hears 750 of these sounds correctly the articulation is rated at 75 %. The writer has used this same scheme for investigating the effects of reverberation and noise upon speech-reception in auditoriums.[1] The "percentage articulation" of an auditorium signifies what percentage of typical speech-sounds can be heard correctly by an average listener in the auditorium. A speaker calls out typical monosyllabic speech-sounds, in groups of three, at a rate of three syllables in two seconds. Observers stationed in representative positions in the auditorium write down what they think they hear. If,

[1] V. O. Knudsen, Phys. Rev., 26, 287, (1925), and 26, 133–138, (1925). See also The Architect and Engineer, 84, 67–72, (1926) and v. 85, (1927).

on the average, they hear correctly four-fifths of the total number of called speech-sounds, the articulation for this auditorium is rated at 80%. It would seem that such a scheme as this offers a satisfactory means for rating the acoustic quality of an auditorium which is to be used primarily for speaking.

It is obvious that the percentage articulation in an auditorium will depend upon (1) the size of the room, (2) the reverberation characteristics of the room, (3) the amount of disturbing noise in the room, and (4) the shape of the room. It is apparent that, for speaking purposes only, the ideal auditorium is a small room free from all noise, and bounded by perfectly absorbing surfaces. In such a small room the listener will be near the speaker and therefore the speaker's voice will be heard with adequate loudness. Further, there will be no interfering noise, reverberation or delayed reflections. Actual tests conducted in a quiet open space have indicated that with average speakers and listeners the articulation in such a room will be about 96%. This figure represents the highest attainable acoustic quality for speech reception in a room. A rating of 100%, that is perfect articulation, can never be attained. A few of the consonantal sounds are sometimes mistaken even under ideal hearing conditions. We are ordinarily unaware of this when we listen to speech because the connotation of the articulated words facilitates the correct interpretations of those words which are not heard distinctly. Even when the speech articulation is as low as 75% the hearing will be regarded as acceptable. An articulation of 96% is, for all practical purposes, about perfect, and therefore there seems to be no necessity for attempting to improve this limited ideal, although it could be done by altering slightly the pronunciation, or even emphasis, of some of the soft consonantal sounds.

The extension of the size of the room, the use of reflecting materials for the walls and ceiling, and the presence of disturbing noise will all tend to impair the acoustic quality of the room, and thus reduce the articulation below 96%. In general, each of the four mentioned factors which affects the acoustics of the room will introduce a distortion or a disturbance which can be determined quantitatively. Thus, the resulting percentage articulation in any specified auditorium can be estimated by the following equation:

$$\text{Percentage Articulation} = 96\, k_l\, k_r\, k_n\, k_s, \qquad (1)$$

where k_r is the reduction factor owing to the inadequate loudness of the speech, k_r the reduction or distortion factor owing to reverberation,

k_n the reduction factor owing to noise, and k_a the reduction factor owing to the shape of the room. The first three of these reduction factors are fairly well known from existing experimental data. The work of Fletcher[2] has determined the effect of loudness upon speech reception, and the work of the writer[3] has determined the effects of noise and reverberation upon speech reception. The results of these determinations, as they affect the problem of hearing in auditoriums, will now be outlined.

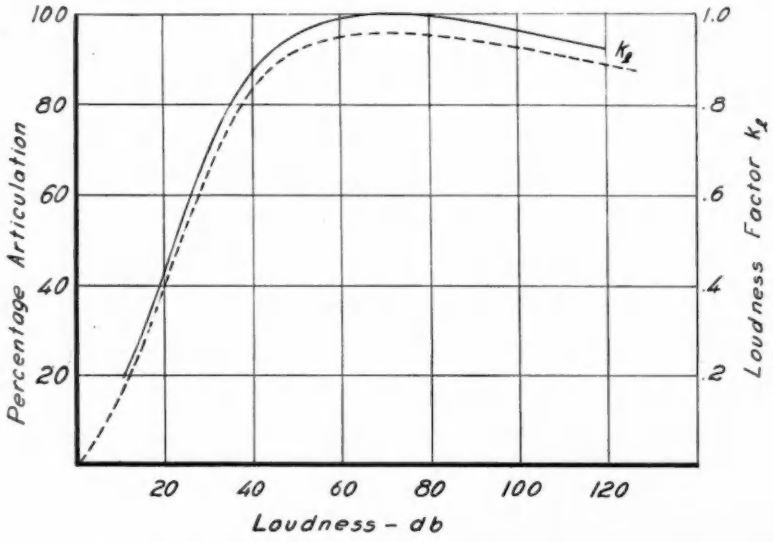

Fig. 1. *Curve showing the effect of loudness upon the reception of speech (taken from Fletcher's data). The dotted curve gives the percentage articulation at different loudness levels. The solid-line curve gives the corresponding loudness reduction factor* k_1.

The Effect of Loudness upon the Reception of Speech in Auditoriums. Fletcher's data on the effect of loudness upon speech reception are shown by the broken line curve in Fig. 1. The loudness of the speech, expressed in decibels (abbreviated db),[4] was controlled by the gain of a distortionless vacuum tube amplifier. It will be seen that the opti-

[2] Nature of Speech and Its Interpretation, Jour. Frank. Inst., 193, 6, (June, 1922).

[3] Loc. cit.

[4] The loudness of a sound in db is ten times the common logarithm of the ratio of the intensity of the sound to the intensity at the minimal threshold of audibility. This unit is becoming universal as the standard unit for the measurement of loudness. It was recently adopted by the Bell System and the International Advisory Committee on Long Distance Telephony in Europe. In recent years, this same unit has received the designation of transmission unit (TU) or sensation unit (SU).

mal loudness of speech appears to be about 70 db, which corresponds to an intensity which is ten million times the intensity which would be just barely audible. This is somewhat louder than normal conversation which is usually of the order of 50 to 60 db, or less. The data represented by Fig. 1 indicate that if the loudness of undistorted speech be between 50 and 100 db, the articulation is above 90%, which is wholly satisfactory. Below 50 db, the articulation drops off rapidly as the loudness is diminished. Thus, at 30 db the articulation is 66% and 20 db it is only 40%.

The solid line curve shown in Fig. 1 has been derived from the broken line curve. This curve gives the value of the loudness reduction factor, k_l, for different loudness levels from 0 to 120 db. The value of k_l at 70 db, the optimal loudness level, is taken as unity, and the value of k_l at all other loudness levels is the ratio of the percentage articulation at that level to the percentage articulation at 70 db. The solid line curve in Fig. 1 is obviously very useful in connection with equation (1), if the average loudness of a speaker's voice in an auditorium be known.

The Average Power of Speakers' Voices in Auditoriums

The data on the effect of loudness upon speech articulation indicate that it would be desirable to maintain the loudness of speech in auditoriums at a level of not lower than 50 db. The question now naturally arises whether the average speaker in an auditorium maintains a loudness level as high as 50 db. It will be seen presently that he does in small rooms, but in large, non-reverberant auditoriums it requires considerable effort on the part of the speaker, and in very large auditoriums it will be impossible to maintain this level without the aid of amplifiers.

The approximate loudness of speech in an auditorium can be determined from simple calculations based upon some numerical constants of speech and hearing obtained by Bell Telephone engineers.[5] The data of Sacia and Sivian at Bell Laboratories indicate that the average speech power generated by an average speaker in normal conversation is about 10 microwatts. The actual power output of different speakers, and even of the same speaker, varies widely from this average value. For example, they found that the peak power may sometimes rise to 2000 microwatts.

[5] H. Fletcher, "Useful Numerical Constants of Speech and Hearing," Bell System Techical Journal, (July, 1925.)

Every public speaker is fully aware that he must raise the intensity of his voice above the ordinary conversational level in order to be heard in a large auditorium. It is evident therefore that his energy output, particularly in very large auditoriums, will be considerably above the average conversational level of 10 microwatts. In order to determine the approximate value of the average power of the average speaker's voice in an auditorium, the writer has obtained some measurements on the loudness of speakers's voices in a small and also in a moderately large auditorium. The measurements were made with the help of a microphone (suspended near the middle of the auditorium), an amplifier with an attenuation circuit and a head-set in its output, and a high quality electric phonograph. The electric phonograph, with a calibrated volume control, was first used for a source of speech in the auditorium. The loudness of the reproduced speech was maintained at different measured levels, and at each level, the attenuation circuit associated with the amplifier (which was located in a remote room) was adjusted until the speech, as heard in the head-set was reduced to the minimal threshold of audibility. A similar adjustment of the attenuation circuit when a speaker was speaking in the auditorium, gave a measure of the loudness of his voice. The method is essentially a substitution method in which the average loudness of the speaker's voice is compared with a measureable loudness from the electric phonograph. The loudness of the speaker's voice is expressed in db. The results obtained from six male speakers (instructors in the University) in the small auditorium are given in Table I. In every case the speaker was unaware that his voice was under observation until after the measurements were obtained.

The loudness of the speaker's voice, measured near the middle of the auditorium, depends principally upon the energy output of the speaker and the total absorption of the room and its contents. It is well established that the average steady state intensity of sound energy I_0 in a room is proportional to the rate of emission of sound energy E and inversely proportional to the total absorption a in the room. The value of I_0 is given[6] by

$$I_0 = \frac{4E}{av}, \tag{2}$$

[6] G. Jaeger, "Zur Theorie des Nachhalls," Acad. Wiss. Wien, Sitzungsberichte, May, 1911, p. 120. (See also E. A. Eckhardt, "Acoustics of Rooms," Jour. Frank. Inst., June, 1923, p. 799. The writer has verified this equation for the steady state of sound energy in a room. See Phil. Mag., 5, 1240–1257, (1928)).

in which v is the velocity of sound in air, about 34300 cm./sec. In eq. (2), E represents, for the present consideration, the average energy output of the speaker, and I_0 represents the average measured intensity of the speaker's voice as "detected" by the microphone and amplifier. In order to evaluate E in microwatts it is only necessary to evaluate I_0 and a in the proper units, and then solve (2) for E. The absorption a should be expressed in sq. cm. of a perfectly absorbing surface, as an open window, and I_0 should be expressed as an energy density, microjoules per cc. The determination of I_0 is based upon the data of Fletcher and Wegel on the sensitivity of the normal ear. Their data indicate that at the minimal threshold of audibility speech has approximately a power flux through each sq. cm. of 9×10^{-10} microwatts. The average energy per cc., assuming a uniform distribution of sound energy throughout the room, would be therefore, at minimal audibility, $9\times10^{-10}/v$, or 2.6×10^{-14} microjoules. If, then, the actual loudness of the speech in the room has been measured in db. it is possible to evalute I_0 in microjoules per cc. by the familiar relation db $=10\log_{10} I_0/I$, where I_m is the energy density at the minimal threshold, that is, 2.6×10^{-14} microjoules per cc. With I_0, a, and V known, and expressed in these appropriate units, the solution of (2) for E will give, in microwatts, the average power output of a speaker. The values of the average power of speakers' voices given in the last columns of Tables I and II were determined in this manner.

TABLE I

Volume of room = 770 cu. meters (27200 cu. ft.).
Absorption of auditorium (empty) = 43.9 sq. meters (472 sq. ft.).

Speaker	Observed Average Loudness*	Total Absorption in Room	Average Power of Speaker's Voice
1 (man)	51.2 db	108.9 sq. meters	32.2 microwatts
1 "	56.2 "	70.6 " "	66.2 "
3 "	50.9 "	82.8 " "	23.0 "
4 "	48.3 "	54.9 " "	8.5 "
5 "	44.7 "	67.9 " "	4.5 "
6 "	53.0 "	66.5 " "	30.0 "
Average	50.7 db		27.4 microwatts

* The average loudness of each speaker was based upon many observations taken during the course of an hour's lecture.

It will be noticed that the different speakers vary between rather wide limits, from 4.5 to 66.2 microwatts. Even the same speaker exhib-

its a wide variation; for in some instances the power would momentarily surge to a peak value of 1000 or 2000 microwatts, and at other times would drop to one microwatt or even less. The average power for the six speakers is 27.4 microwatts. It is admittedly impossible to obtain an accurate determination of the normal power of the average speaker in an auditorium from so few as six speakers. However, the average value obtained from these measurements is certainly a good approximation, and is sufficiently representative of the normal value to serve a useful purpose in the present investigation. On the basis of these few measurements, it appears that in a small auditorium (770 cu. meters) the average speaker increases the power of his voice about 170% above the power level of ordinary conversation, which telephone engineers have found to be about 10 microwatts.

A similar series of measurements on the power of speakers' voices has been conducted in a larger auditorium. The results of these measurements are given in Table II.

TABLE II

Volume of room = 6790 cu. meters (240,000 cu. ft.).
Absorption of auditorium (empty) = 335 sq. meters (3600 sq. ft.).

Speaker	Observed Average Loudness	Total Absorption in Room	Average Power of Speaker's Voice
1 (man)	49.4 db	335 sq. meters	65.5 microwatts
2 "	45.6 "	335 " "	27.5 "
3 "	46.1 "	413 " "	37.8 "
4 (woman)	43.0 "	632 " "	28.4 "
5 (man)	43.5 "	531 " "	26.8 "
6 "	51.0 "	502 " "	142.0 "
7 "	42.7 "	560 " "	23.4 "
8 (woman)	44.3 "	629 " "	38.2 "
Average	45.7 db		48.9 microwatts

Again, it will be noticed that the different speakers vary considerably in their speech-power, from 23.4 to 142.0 microwatts. As would be expected, the average speaker speaks with greater energy in a large auditorium than he does in a small one. Thus, the average power of the speaker's voice in the large auditorium is 48.9 microwatts compared with 27.4 microwatts in the small auditorium. However, the increase in the average power of a speaker's voice in a large auditorium is not sufficient to compensate for the diminution of loudness owing to the greater amount of absorption in the large auditorium. Thus, the observed average loudness of six speakers in the small auditorium was 50.7

db, whereas the average loudness for eight (other) speakers in the large auditorium was only 45.7 db. It will be noticed by referring to Fig. 1, that a loudness of only 45.7db is approaching a dangerously low level A slight disturbance from noise or reverberation, or a slight "fading" of the voice, will result in unsatisfactory hearing.

The early Greeks were fully aware of this inadequacy in the loudness of speakers' voices, and attempted to compensate for it, especially in the larger theatres, by two different devices. The actors on the stage wore huge masks which not only exaggerated facial expressions so that they could be seen from the most distant seats but also served to enhance the loudness of the voice by reason of the shapes of the masks which incorporated the principle of the megaphone. In addition, we are informed in the writings of Aristoxenus, a large number of bronze vessels fashioned into resonators, were distributed in regularly spaced niches throughout the theatre. In large theatres there were three horizontal ranges of resonators at equally spaced vertical levels, with twelve resonators in each range. These resonators were all carefully tuned to resonate to the various notes of musical systems, for the purpose of emphasizing the more important frequency components of speech and music. One range of resonators was tuned for the anharmonic, another the chromatic, and the third for the diatonic system. The actual merit of these resonators for enhancing the loudness and pleasing qualities of speech and music is rather difficult to assess, but it is doubtful that they were of any real value. On the other hand, the combined mask and megaphone was of unquestioned value for augmenting the loudness of the voice—at the same time, however, distorting the natural quality of the voice.

The use of these two devices—the megaphone and the resonator—most clearly indicated that in their open air theatres the Greeks were handicapped by the same difficulty that was revealed by the recent investigation of the loudness of speech in auditoriums; namely, that the natural, unaided voice does not provide an adequate supply of speech energy for good hearing in large auditoriums.

Effect of Reverberation upon the Reception of Speech in Auditoriums

Fig. 2 shows how speech articulation depends upon the time of reverberation[7] in a group of auditoriums having about the same shape

[7] The time of reverberation is here used as defined by W. C. Sabine, that is, the time required for the intensity of a tone of 512 d.v. to decay to one-millionth of its initial intensity.

and volume (200,000 to 300,000 cu.ft.) but different times of reverberation. The small circles in Fig. 2 show the observed values of percentage articulation for the corresponding measured times of reverberation in the auditoriums investigated in this series. The lower curve is drawn to represent the most probable fit with the observed data. It will be noted that, approximately, the articulation decreases 6% for each additional second of reverberation.

The data for the three auditoriums having times of reverberation less than 2.0 seconds, and also the data for six other auditoriums, have been

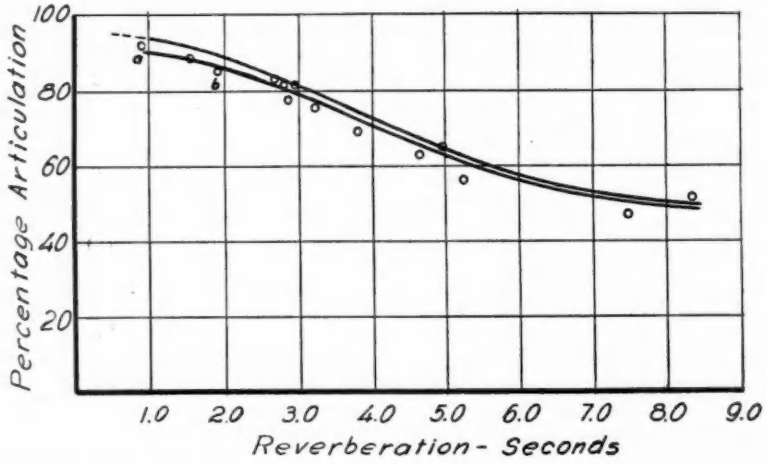

Fig. 2. *Curves showing the interfering effect of reverberation upon the hearing of speech. The lower curve represents the most probable fit with the observed data. The upper curve has been corrected for loudness, and corresponds to a loudness of 70 db.*

obtained during the past three years and therefore were not included in earlier publications. The new data represented in Fig. 2 seem to strengthen the conclusion suggested in the earlier papers, that the optimal time of reverberation for the reception of speech in an auditorium is somewhat shorter than is attained or planned in current practice. The optimal time of reverberation, based upon the combined effects of loudness and reverberation, will be given consideration in a later section in this paper.

The lower curve in Fig. 2. which represents the mean result of the experimental determinations, was not obtained for a constant loudness of speech, because the loudness is dependent upon the amount of absorption in the room. Assuming the power of the speaker's voice to

remain constant,[8] the resulting intensity of the speech would be almost inversely proportional to the total amount of absorption in the auditoriums, or directly proportional to the time of reverberation. It was found experimentally that the average loudness of the speakers' voices used in these tests, in an auditorium having a volume of 8440 cu. m. (300,000 cu. ft.) and a time of reverberation of 1.50 seconds, was about 48 db. Using this datum, and the loudness-articulation data given in Fig. 1, it is possible to correct the lower curve in Fig. 2 for variation of loudness. The upper curve in Fig. 2 was obtained by applying such a correction so as to give the percentage articulation for a uniform loudness level of 70 db, which is the loudness level for optimal hearing. This curve has been extrapolated to a time of reverberation of .50 second, as indicated by the dotted portion of the curve. Such an extra-polation is warranted by articulation tests the writer has conducted in a small room in which the percentage articulation increased as the time of reverberation was reduced from 1.0 to .60 seconds.

It is now possible, from the upper curve in Fig. 2, to derive k_r, the reduction factor owing to reverberation, for times of reverberation between .5 and 8.0 seconds. The value of k_r is taken as unity for a time of reverberation of .5 second. The value of k_r for any other time of reverberation is the ratio of the articulation at that time to the articulation for a time of .5 second. The curve in Fig. 3 gives the value of k_r, obtained in this manner, for different times of reverberation. It will be noted that k_r decreases almost uniformly as the time of reverberation increases from 1.0 to 6.0 seconds. Above 6.0 seconds the rate of decrease of k_r appears to be less rapid.

The departures of the experimentally determined "percentage articulations" from the values indicated by the smooth curve in Fig. 2 are greater than the experimental errors in determining the time of reverberation and the "percentage articulation." These departures are more probably the result of other factors affecting the reception of speech, such as residual noise and the shape of the room. The tests were conducted during the quiet part of the night, but in many instances there were slight disturbances from passing automobiles or trolley cars. Further, although care was exercised in the selection of auditoriums which would be nearly uniform in shape and size, there

[8] This assumption seems more plausible than the alternative one that the speaker maintains a constant loudness level. It seems likely, however, that neither assumption is correct. A speaker generally attempts to raise his voice to the loudness level required for satisfactory hearing, but is limited by the physical characteristics of his speech apparatus.

were certain variations in length, width and height, and in architectural detail. Finally, there are other variations, attributable to the reverberation characteristics of the rooms, which are not included in the independent variable (the time of reverberation) used in Fig. 2. Thus, the dependence of reverberation on pitch and the location of the obsorptive materials in the room both affect the hearing properties. The importance of these two aspects of reverberation has not yet been quantitatively determined, but is under investigation at the present time. Most auditoriums, especially when filled with auditors, are very much more reverberant for the low pitched tones than for the high ones. For example, one of the auditoriums studied

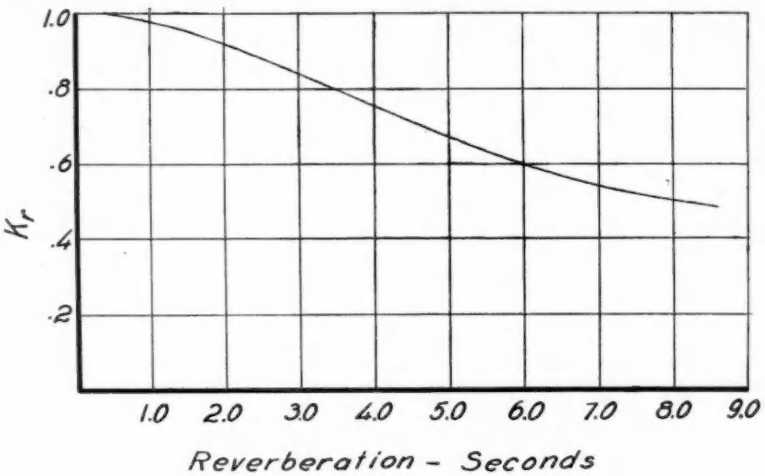

Fig. 3. *Curves giving the reverberation reduction factor k_r for different times of reverberation.*

in this investigation (the one marked (b) in Fig. 2) had a time of reverberation, empty, of slightly less than two seconds for a tone of 512 d.v., and nearly five seconds for a tone of 128 d.v. The determined "percentage articulation" for this auditorium falls slightly below the smooth curve. On the other hand, another auditorium[9] (the one marked (a) in Fig. 2) had a time of reverberation of .94 second at 512 d.v. and 1.10 seconds at 128 d.v. This auditorium exhibits an un-

[9] This auditorium is a sound studio designed by the writer for Metro-Goldwyn-Mayer. The entire ceiling and walls are lined with multiple layers of fibre building board and loose wool separated by air-spaces. This type of absorptive treatment gives an almost uniform time of reverberation for tones of all pitch. The acoustic properties of this room are regarded by the writer and by those who use it as practically perfect, not only for speaking but also for the recording of both speech and music.

usual instance of nearly uniform reverberation for tones of different pitch. The determined articulation for this auditorium lies slightly above the smooth curve of Fig. 1. It is probable that the departures of the points (a) and (b) from the smooth curve in Fig. 2 are actually attributable to the manner in which the reverberation varies with pitch. It is to be noted though, that since these departures are small, it is not likely that the development of absorptive materials that are uniformly absorptive for tones of all pitch will produce a great improvement in the speech-reception properties of rooms. The need for such uniformly absorptive materials appears to be desirable however for naturalness in the quality of speech, and more especially music.

Another characteristic of reverberation in a room is caused by the distribution of the absorptive material used in a room. Watson[10] recently has advocated the use of a highly absorptive space for the listener and fairly reverberant space for the generation of sound. That is, the stage would be left sufficiently reverberant to satisfy the taste of musicians and speakers, and the main part of the auditorium would be treated with very absorptive materials so that there would be little, if any, interfering effect from reverberation. As yet, no quantitative tests have been conducted to determine the effect on speech articulation of such a proposed distribution of absorptive material in an auditorium. The effect, no doubt, is beneficial, especially in large auditoriums where the reflected sound from the stage surroundings ("sounding board") enchances the direct sound wave.

Both of these features just mentioned—the variation of reverberation with pitch and the distribution of absorptive material in the auditorium—will have an effect upon the reduction factor k_r. However, it is not probable that either of these features has a very significant effect upon the "percentage speech articulation" in an auditorium. It is probable therefore that for a first approximation, the value of the time of reverberation for a tone of 512 d.v., as is commonly employed in current practice, can be used for determining, by means of Fig. 2, the appropiate value of k_r for any auditorium. The two features of reverberation under discussion are of unquestioned value in auditorium design, but their most important significance is in relation to the preservation of naturalness of speech and music rather than the improvement of "speech articulation."

[10] F. R. Watson, "Ideal Auditorium Acoustics," Jour. of Am. Inst. of Architects, July, 1928.

Effect of Noise upon the Reception of Speech in Auditoriums

It is a common observation that noise interferes with the hearing of either speech or music; that it produces what is called a "masking effect," the magnitude of which depends upon the loudness of the noise compared with the loudness of the speech or music. The interfering effect of noise upon the reception of speech in auditoriums has been investigated by the writer a number of years ago[11]. The principal

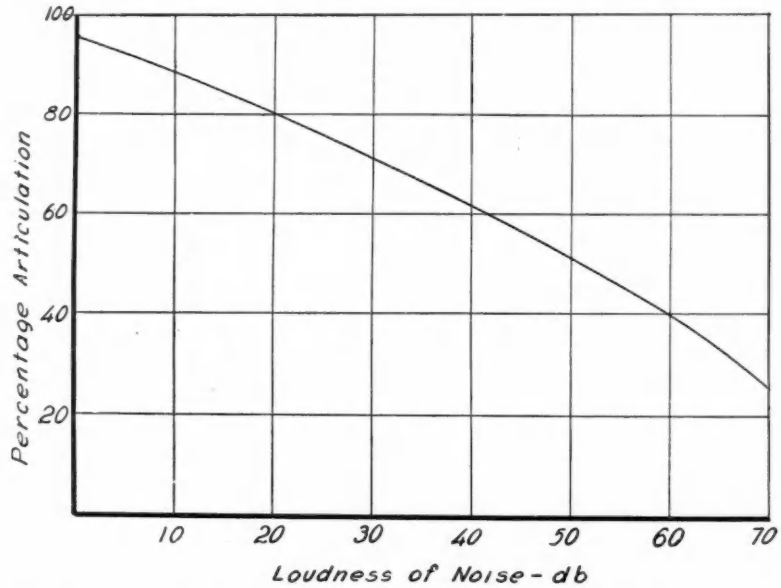

Fig. 4. *Curve showing the interfering effect of noise upon the hearing of speech.*

result of that investigation as applied to the present problem, is represented by the curve in Fig. 4. This curve shows how the articulation is affected by an interfering noise of different loudness levels, for speech which was maintained at a level of 47 db. The tests were conducted in a room which had a volume of 422 cu. m. (15000 cu. ft.) and a time of reverberation of 1.3 seconds. Since the noise tests were conducted in a room having a time of reverberation of 1.3 seconds, there was an interfering effect from reverberation, the magnitude of which can be determined from the curve in Fig. 3. The curve in Fig. 4

[11] V. O. Knudsen, "Interfering Effect of Tones and Noise upon the Reception of Speech," Phys. Rev., 26, 133–138, (1925).

has been corrected for the excess of reverberation above .5 second, and thus the ordinates in this curve are slightly greater than the observed values of the percentage articulation.

The interfering noise was conducted to the observer's ears by means of a pair of telephone receivers which were adjusted on the headband so that each receiver was maintained at a fixed distance of about one inch from the ear. In this manner, the listener observed the speech in the presence of a controllable noise, and further, the speaker was not bothered by the interfering noise. It will be noted that the artic-

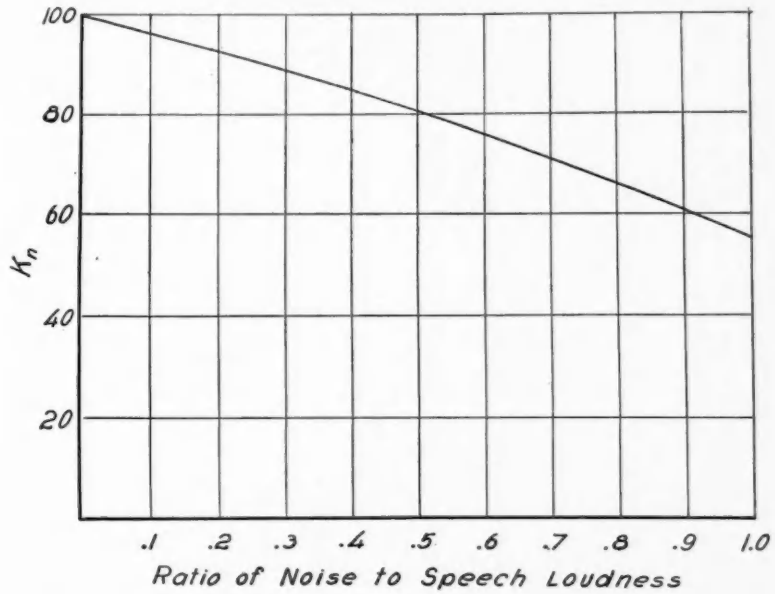

FIG. 5. *Curve showing how the noise reduction factor k_n depends upon the loudness of noise. The abscissa gives the ratio of the noise loudness to the speech loudness, where both are expressed in db.*

ulation decreases almost uniformly as the loudness of the noise increases. Further, it will be seen that even a slight noise produces an appreciable impairment. The complete absence of noise is thus seen to be an important factor for ideal hearing.

The harmful effect of actual noises in auditoriums is indicated by some tests which were conducted in two different auditoriums. In a certain high school auditorium, about which the writer was consulted, it was found that one of the chief defects was caused by noise from the ventilating fan and motor. Speech tests conducted in the empty

auditorium, with the fan and motor in operation, showed an average speech articulation of 55%. With the fan and motor shut down the average articulation increased to 68%. In another auditorium it was found that the noise from an adjacent corridor was a source of interference. The separation of the auditorium from the corridor by means of a vestibule increased the articulation in the rear portion of the auditorium from 29.7 to 54.5%.

The curve in Fig. 5 is derived from the curve in Fig. 4. It gives the value of k_r for different loudness levels of noise. The loudness of the noise is here represented by the ratio of the noise, in db, to that of the speech, also in db. Thus, when the noise is at the same loudness level as the speech, the abscissa in Fig. 6 is 1.0. The value of k_n for no noise is taken as unity, and all other values of k_n are obtained by taking the ratio of the ordinate (in Fig. 4) for the loudness level in question to the ordinate for zero noise. This method of determining k_n is not strictly rigorous but it gives a close approximation which is sufficiently accurate for practical problems in auditorium acoustics.

The manner of using this curve would be as follows: First determine, by measurement if necessary, the average loudness of the noise in the auditorium. Take the ratio of this loudness level, in db, to the probable loudness level of the speech, in db, and read off from the curve in Fig. 5 the appropriate value of k_n. Thus, if it is found that the average noise level in an auditorium is 10 db and the average speech is 50 db, the value of k_n would be .925. The average noise prevalent in typical auditoriums is rarely lower than 5 db, and may sometimes be as high as 20 to 25 db.

Effect of Shape of Auditorium upon Speech Reception

The auditoriums in which the writer has investigated the effect of reverberation and noise upon the reception of speech were typical school and theatre auditoriums, and were essentially rectangular in shape, having dimensions of about $80' \times 110' \times 32'$ high. No quantitative tests have, as yet, been conducted in auditoriums which differ only in shape. There is undoubtedly some benefit to be gained from the use of suitably designed sounding boards, or from suitably located wall and ceiling surfaces near the speaker, but more data are yet needed to decide the exact benefit derived from such devices. The writer has tested the effect of a large reflecting sheet iron surface placed directly behind the speaker. Speech articulation tests were conducted in an

empty auditorium with and without the reflector. The reflector was 10′ high and 12′ wide, and was placed directly behind the speaker, who stood near the middle of the stage floor. With the reflector in position, the average articulation in the auditorium was 62%, and with the reflector removed it was 59%. The observed difference of 3% is not great but it is greater than the experimental error and the reflector therefore is shown to possess some benefit. A similar benefit was observed, with the speaker on the front of the stage, when the

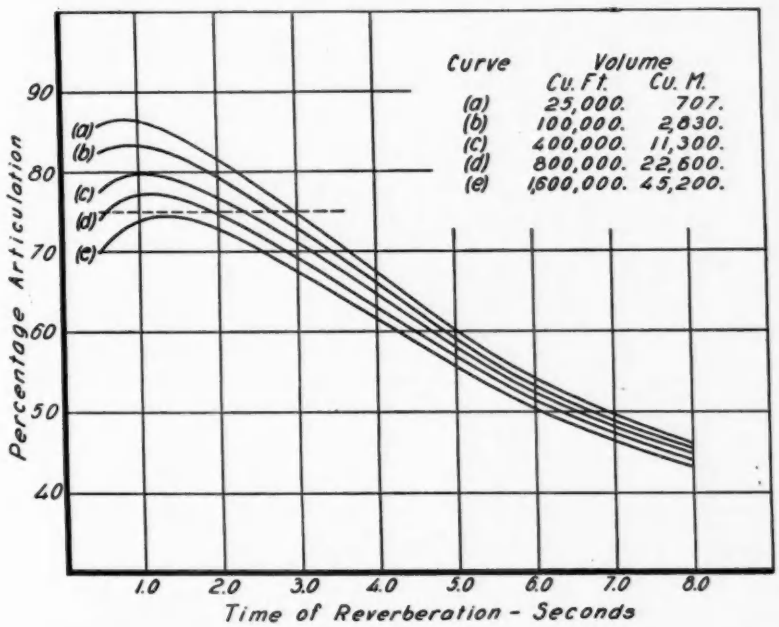

FIG. 6. *Group of curves giving the probable percentage articulation in auditoriums of different sizes and with different times of reverberation. These curves indicate that there is an optimal time of reverberation for the hearing of speech in an auditorium of a certain size.*

asbestos curtain was lowered so as to serve as a reflecting surface directly behind the speaker.

The influence of the shape of an auditorium upon speech reception requires further quantitative investigation. In the auditorium of conventional rectangular shape, it is probable that the k_s (as used in eq. (1)) does not differ appreciably from 1.0. In very large auditoriums, especially with curved surfaces, it is probable that k_s may be reduced to a value as low as .90. It is possible that in small rooms, or in auditoriums designed with properly shaped and located reflecting surfaces,

k_s may reach a value as high as 1.05. For practical guidance in the design of auditoriums, it probably is advisable to assign a value of 1.0 to k_s, unless the shape of the auditorium is of peculiar design.

Combined Effects of Loudness and Reverberation upon the Reception of Speech in Auditoriums

In the earlier sections of this paper the effects of loudness and reverberation upon speech reception were considered separately. It is obvious that as the time of reverberation in an auditorium is reduced, the average loudness of speech, assuming a constant power rate for the speaker, will be reduced correspondingly. This suggests that there may be an optimal time of reverberation for speech in an auditorium. This would occur when a further reduction in the reverberation would

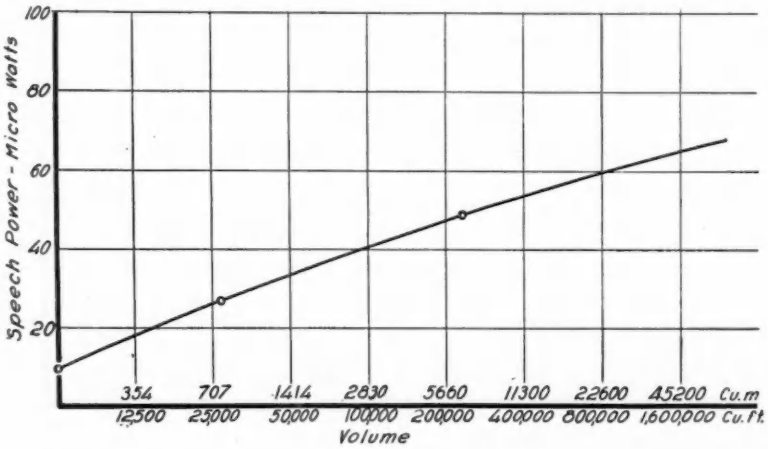

Fig. 7. *Curve showing the probable speech power of the average speaker in auditoriums of different sizes.*

concurrently reduce the loudness to the extent that the impairment occasioned by the diminished loudness would just compensate for the improvement occasioned by the reduction of the reverberation. The manner in which this occurs is indicated by the series of curves which are plotted in Fig. 6. These curves give the calculated values of the percentage articulation in auditoriums of different sizes and times of reverberation, for the probable average loudness of speech of the average speaker, based upon the measurements made in this investigation. The average speech-power of the average speaker in auditoriums of different sizes is obtained from the curve in Fig. 7, which

is based upon data obtained in the two auditoriums studied in this investigation and also upon the datum given by Bell Telephone engineers, namely that the average speech-power of the average speaker in a small room is 10 microwatts.[12] Having determined, from Fig. 7, the probable speech-power of the average speaker in a room of a certain size, and assuming that the speaker maintains this power output for different times of reverberation,[13] it is possible to calculate the resulting loudness in an auditorium of any size or time of reverberation. A typical set of calculations for an auditorium is given in table III. The values of k_l and k_r are determined from the curves

TABLE III

Volume of auditorium = 11,330 cu. meters (400,000 cu. ft.).
Average speech-power (from Fig. 7) = 54 microwatts.

Time of Reverberation	Average Speech Intensity*	Average Loudness, db	k_1	k_r	$k_1 k_r$
.50	$.665 \times 10^4$	38.2	.850	1.00	.850
.75	1.00	40.0	.874	.993	.868
1.00	1.33	41.2	.885	.982	.870
1.25	1.66	42.2	.894	.967	.865
1.50	2.00	43.0	.900	.953	.858
2.00	2.66	44.3	.910	.924	.840
3.00	4.0	46.0	.925	.837	.775
4.00	5.3	47.3	.936	.752	.704
6.00	8.0	49.0	.950	.600	.570
8.00	10.6	50.3	.959	.510	.489

* The speech intensity is given in terms of the minimal audible speech intensity.

in Figs. 1 and 3 respectively. It is assumed that the auditoriums are of the typical rectangular shape, so that $k_s = 1.0$. It also is assumed that the rooms are relatively free from disturbing noise, so that the residual noise is only one-tenth as loud as the speech, and therefore k_n will be .96. Eq. (1) then becomes

$$\text{Percentage Articulation} = .922\ k_l\ k_r.$$

The ordinates of the curves in Fig. 6 were calculated by the use of this equation and a series of tables like Table III.

[12] It is assumed that this speech-power will be maintained by the average speaker in a room having a volume of 175 cu. meters (6200 cu. ft.).

[13] It is possible that a speaker raises the intensity of his voice as more and more absorption is brought into a room. This has not yet been tested, but it is not likely that a large error is is introduced by the assumption of constant power of the speaker's voice in the same room.

It is obvious that, for an auditorium of a certain size, the optimal time of reverberation will be the time for which the product $k_l k_r$, or the percentage articulation, will be a maximum. Thus the maximal values of the ordinates in the group of curves in Fig. 6 indicate the optimal times of reverberation in auditoriums of different sizes, for speech of about the average loudness that would be commonly used.

The entire series of curves shows very clearly how the reception of speech depends upon the size and time of reverberation of an auditorium. As would be expected, the optimal time of reverberation for speech reception in a small room is as short as .75 seconds. In a very small room the loudness is adequate and therefore speech will be heard more clearly and distinctly the nearer the reverberation approaches zero. In large auditoriums a somewhat longer time is advantageous because it promotes loudness. It will be noted that the peaks of the curves in Fig. 6 are rather broad and flat. This would seem to indicate that there is a considerable allowable variation in the time of reverberation from the optimal time, without appreciable sacrifice in hearing quality. In the design of auditoriums, therefore, it is desirable to so choose the absorptive treatment of the auditorium that the absorption furnished by different sized audiences will make the time of reverberation vary between the limits which determine the approximately flat portion of the curve.

Another factor must also be considered, namely, that the optimal time of reverberation for music is somewhat longer than for speech, and therefore it would be desirable to compromise between the requirements for speech and for music, especially in auditoriums which are to be used both for speech and music.

The optimal time of reverberation for auditoriums has been determined by Watson,[14] Lifshitz,[15] and others. These investigators have arrived at the optimal time primarily by calculating or measuring the time of reverberation in auditoriums which are pronounced good by competent critics. Lifschitz has derived a semi-empirical formula for calculating the optimal times of reverberation[16] for auditoriums of different sizes. This formula yields results which are in fair agreement

[14] F. R. Watson, "Acoustics of Buildings," John Wiley and Sons; Jour. Frank. Inst., July, 1924.

[15] Samuel Lifschitz, Phys. Rev., 27, 618–621, (1926).

[16] This formula is based upon the assumption that the product of the time of reverberation and the loudness, expressed in db, should be constant. Lifshitz calls this product "the energy of musical perception." The assumption seems rather arbitrary but it is fairly consistent with the observed facts.

with Watson's results and with currently accepted optimal times of reverberation. The top curve in Fig. 8 shows the values of the optimal time of reverberation as given by Lifschitz. Although Lifschitz states that his results apply both to speech and music, it is probable that they apply more particularly to music, since the results are based upon the judgments of listeners who regard loudness, resonance, euphony and other qualities as determining factors. The lowest curve in Fig. 8 shows the optimal time of reverberation for speech, based upon the maximal values of the curves in Fig. 6. It would seem that the bottom and top curves in Fig. 8 give the most trustworthy available data for determin-

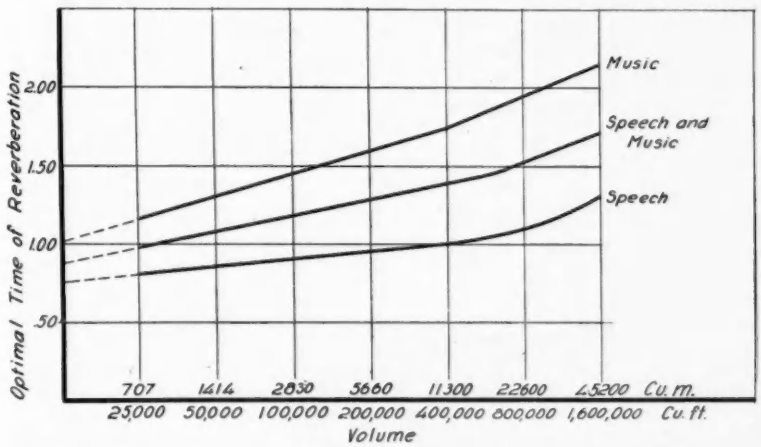

FIG. 8. *Curves showing the optimal time of reverberation for auditoriums of different sizes. The upper curve is taken from the data of Watson and Lifshitz. The lower curve, for speech, is obtained from the maxima in Fig. 6. The middle curve is the arithmetical means of the upper and lower curves, and represents a reasonable choice for both speech and music.*

ing the optimal time of reverberation in auditoriums for either speech or music, where no provision is made for amplifying the power of the voice. If an auditorium is to be used both for speech and music, as is usually the case, it would seem advisable to use the mean value of the two curves. The middle curve is such a mean value curve, and thus gives the optimal time of reverberation for both speech and music.

The importance of the loudness of speech in a large auditorium is strikingly shown by the curves in Fig. 9. These curves have been calculated to show especially how the loudness of speech affects the hearing intelligibility in an auditorium having a volume of 11,300 cu. m. (400,000 cu. ft.). The curve marked (f) is for weakest voice of the fourteen speakers used in this investigation. It will be seen that the arti-

culation for such a speaker, even under the most favorable listening conditions in this auditorium, does not exceed 70%. On the other hand, the curve marked (b) is for the loudest speaker in this series. The curve marked (e) is for the average of the four weakest speakers and the curve marked (c) is for the average of the four loudest speakers tested in the large auditorium, The curve marked (d) is for the average of the eight speakers. It will be noticed that the louder speakers are

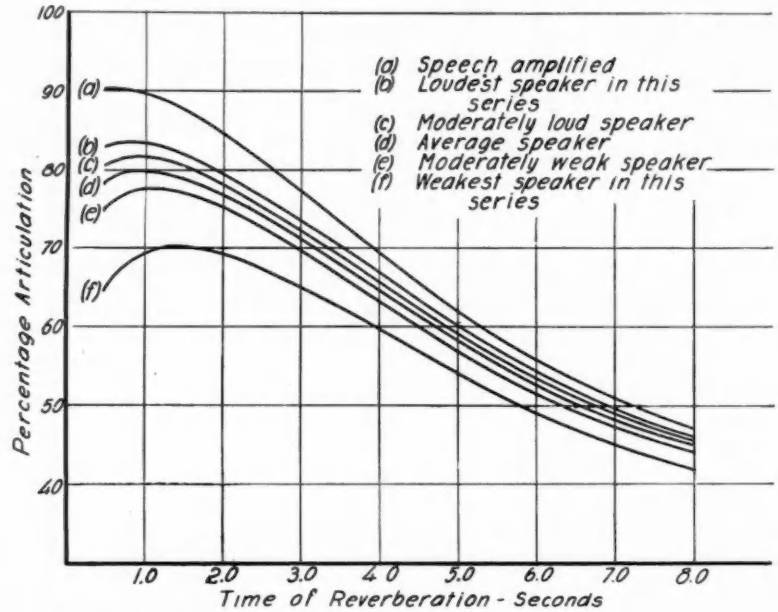

Fig. 9. *Group of curves showing how the loudness of a speaker's voice effects the hearing of speech in auditoriums. These curves are for an auditorium having a volume of 400,000 cu. ft. The loudness of a speaker's voice is seen to be an important factor.*

heard very much better than the weaker ones—the optimal articulation for the loudest speaker is 83.5% as compared with 70% for the weakest one. This difference is quite significant inasmuch as an articulation of 75% is required for satisfactory hearing.

It will be noted further, by referring to the curves in Fig. 9, that the optimal time of reverberation is different for different speakers even in the same auditorium, varying from about .85 seconds for the loudest speaker to 1.40 seconds for the weakest speaker. However, a time of reverberation of about 1.00 to 1.25 seconds will quite satisfactorily approximate the optimal reverberation for all speakers.

Curve (a) in Fig. 9 has been calculated upon the assumption that the speech has been amplified, without distortion, to an energy level

corresponding to a loudness of 60 db in this same auditorium when the time of reverberation is 1.0 second. The advantage of such distortionless amplification of speech is clearly indicated. Thus, with a time of reverberation of less than 1.0 second, the articulation is 90%, which could be regarded as practically perfect for the hearing of speech. These calculations seem to indicate that suitable amplifiers (public address systems) are imperatively needed in large auditoriums. In addition, such amplifiers are also beneficial in smaller auditoriums, especially if the auditorium is beset with disturbing noise.

At a loudness level of 60 db. there is no necessity of reverberation for the usual purpose of enhancing the loudness. In fact, the reverberation should be kept as low as is consistent with other considerations, such as maintaining sufficient brilliance and resonance in the room to meet the requirements for music. Curve in Fig. 9 indicates that the time of reverberation should not exceed 1.0 second for speech. Even for music, there seem to be no physical factors which would warrant a time of reverberation much in excess of 1.0 second (which is about the optimal reverberation for speech and music in a small room) if the loudness be maintained at about 60 db.

The present public address systems ordinarily introduce a certain amount of distortion because of the limitations of the electrical and acoustical equipment used, and therefore a distortion factor k_d should be introduced in eq. (1) when such amplifiers are used. In a properly designed public address system the value of k_d probably will be no less than .95. This point, however, requires further investigation.

Referring again to Fig. 9, and recalling that curve (a) was based upon distortionless amplification, it will be noted that if the amplifier introduces considerable distortion, the amplifier may be a hindrance rather than an aid to better hearing. Thus if the distortion factor k_d be less than .90, the added distortion will more than offset the advantage gained from increased loudness for all speakers except those with weak voices. It is important therefore that public address systems for auditoriums be of the high quality type.

The curves in Fig. 6 are of considerable value in placing a quantitative estimate on the acoustic merit of different auditoriums. They also indicate the limits of size and reverberation which can be tolerated if the percentage articulation is to be maintained at a satisfactory level. Experience has shown that if the average articulation in an auditorium be 75% or more, the hearing conditions are regarded as satisfactory. It is possible to understand speech when the articulation is as low as 65%,

but it requires normal acuity of hearing and strained attention. If it be desired to keep the articulation above 75%, and it seems to the writer that 75% should be regarded as the admissible minimum, the size and time of reverberation of the auditorium are limited to values above the broken horizontal line in Fig. 6. Thus, it would seem advisable to regard about 800,000 cu. ft. (22600 cu. meters) as the upper limit to the size of an auditorium which is to be used for speaking, unless some amplifying equipment be installed for increasing the loudness. It should be borne in mind that this limitation is based upon the requirements for the average speaker. For speakers with moderately weak voices (see Fig. 9), the size should not exceed 400,000 cu. ft. (11,330 cu. meters); and for speakers with very weak voices, the size should not exceed 100,000 cu. ft. (2830 cu. meters).

The admissible limits of the time of reverberation, in order that the speech articulation be above 75%, are also indicated plainly in Fig. 6. Thus, in an auditorium having a volume of 400,000 cu ft. the time of reverberation should not exceed 2.4 seconds. The 2.4 seconds should be regarded as the upper admissible limit for the time for reverberation in such an auditorium when it is used with the smallest probable audience it is to accommodate. It is good practice to design an auditorium such that this upper admissible reverberation is obtained with no audience present, and that the auditorium have the optimal time of reverberation with the most probable sized audience present.

The principles and results described in this article abviously can be used to good advantage in the design of new auditoriums. To illustrate, the actual problem in a typical large lecture hall will be worked out, using the English units only. The dimensions of the lecture hall are $50' \times 80' \times 25'$—the volume of the room is therefore 100,000 cu. ft.—and the hall is to accommodate an audience of 540 persons (all adults), with the expectation that the most probable number of auditors in the room will be 360 persons. An inspection of Fig. 8 will show that the optimal time of reverberation for speaking in this hall is .90 seconds. The total number of units of absorption, a, required to furnish this optimal reverberation is given by[17]

$$a = \frac{.048 \times 100000}{.90} = 5333 \text{ sq. ft. units.}$$

[17] The factor .048 may be called the reverberation constant. A value of .05 is often used, but the experience of many investigators indicates that .05 is too large, especially in rooms which have irregular shapes. The factor obviously is proportional to what might be called the "mean free path," that is the average distance between successive reflections of sound waves in the room.

Suppose the floor is cork linoleum having a coefficient of sound-absorption at 512 d.v. of .05. Upholstered seats, each supplying an absorption of 1.80 units, are to be installed, in order, among other things, that the hall be not too reverberant when only a few auditors are present.

The total absorption supplied by the most probable sized audience, the unoccupied seats, and the floor will then be:

360 persons at 4.2 units each.....	1512 sq. ft. units
180 upholstered seats at 1.8 units each	324 sq. ft. units
4,000 sq. ft. of floor at .05	200 sq. ft. units
TOTAL.....	2036 sq. ft. units.

It is necessary therefore that the walls and ceiling supply 5333—2036, or 3297 sq. ft. units. This can be attained by the use of one or more of many available acoustic materials for the interior finish of the walls and ceiling. An effective treatment would consist of (1) treating the entire ceiling (4000 sq. ft.) with an acoustic tile having a coefficient of .50 at 512 d. v., which would supply 2000 units; and (2) treating the walls down to three feet from the floor, (i. e. 5720 sq. ft.) with an acoustic plaster having a coefficient of .21 at 512 d. v., which would supply 1201 units. The remaining deficiency of 96 units would be approximately supplied by a hard plaster wainscot and miscellaneous furnishings in the room.

The times of reverberation and the probable percentage articulation in the room for different sized audiences, assuming the room to be reasonably free from noise, will be as follows:

No. of Auditors	*Time of Reverberation*	*Percentage Articulation*
None	1.08 sec.	83.6
180	.98 "	83.8
360	.90 "	83.9
540	.83 "	83.9

The lecture hall, thus treated, has an articulation of nearly 84% for either small or capacity audiences, which will provide highly satisfactory hearing conditions. If the walls and ceiling of the lecture hall had been finished with ordinary hard plaster, and the seats had been of the usual wood type, the following hearing conditions would have resulted.

No. of Auditors	*Time of Reverberation*	*Percentage Articulation*
None	8.05 sec.	45.9
180	3.62 "	69.5
360	2.22 "	79.1
540	1.66 "	81.8

Under these conditions the hearing would be acceptable when more than 300 persons are in the room, but with a small audience the hearing would be altogether intolerable.

ERRORS IN THE RECOGNITION OF SPEECH SOUNDS IN AUDITORIUMS

In the preceding sections the quality of hearing in auditoriums has been rated in terms of the percentage articulation. In a study of the hearing of speech in auditoriums the nature of the hearing errors, as well as the number of errors, is a matter of considerable interest. For example, a knowledge of the nature of the errors in the recognition of speech in auditoriums should prove to be useful in the teaching of oral expression. In one of the auditoriums investigated in this series, the errors have been tabulated for the purpose of determining the frequency of occurrence of the different errors in the recognition of speech. For example, of the total of 2200 monosyllabic speech-sounds called out in this auditorium, 1213, or 55.1%, were recognized correctly. The total number of consonants in these 2200 speech-sounds was 4048, of which 3093, or 76.4%, were recognized correctly. Similarly, of the total 2200 vowel sounds, 2052, or 93.3%, were recognized correctly. The total number of consonant errors was 955. Of these, 529 were final consonants, 224 were initial consonants, 117 were omissions of the called consonant, and 85 were additions which were not called. The preponderance of the errors among final consonants is noteworthy, and is probably the result of the masking produced by the reverberation of the preceding vowel.

A complete tabulation of the consonantal errors in this one auditorium is given in Table IV. The first column gives the consonant called; the second gives the number of times it was called. The next 24 columns give the classification of errors—"Omission" meaning that it was erroneously omitted and "Addition" meaning that it was erroneously added. For example, *d* was heard as *b* 14 times, as *g* 22 times, was omitted 19 times, was added 12 times, but was never heard as *ch s, w* or *y*; *ng* was heard as *m* 21 times, as *n* 41 times,was omitted 14 times

but was never added (in error) or heard as *ch*, *f*, *h*, etc. The last two columns give the total number of errors and percentage of error for each consonant. It will be seen that *ng* was mistaken more frequently than any other sound—94 times out of 176, or 53.4% of the total number of times it was called. Similarly, *d* was mistaken 42.7% and *v* 39.2% of the total number of times they were called. On the other hand, such consonants as *r*, *s*, *w*, and *y* are quite rarely mistaken. Thus, *y* was mistaken only 5.1% and *w* only 6.2% of the total number of times they were called.

TABLE IV

SOUNDS CALLED	NO. OF TIMES HEARD	CLASSIFICATION OF ERRORS																						TOTAL NO. OF ERRORS	PERCENT OF ERRORS
		b	*ch*	*d*	*f*	*g*	*h*	*j*	*k*	*l*	*m*	*n*	*ng*	*p*	*r*	*s*	*sh*	*th*	*t*	*v*	*w*	*y*	*z*	OMISSION		
b	220		1	6	1	6	2		2	1	3			1		1	2			12	2			5	5	52	23.6
ch	176								5									3	1	4					1	14	8.
d	220	14			1	22	1	1	1	1	1	4	2		2					4	9				18	12	94	42.7
f	176	10					2		6		1	1		4		1		13	7	11					8	4	68	38.6
g	176	8	1	11				2		3	1	2	5		3				1		4		1	1	14	5	62	35.2
h	176	1			1				4					2						2					14	4	28	15.9
j	176		7	4		17											4	1		1	1	6	3	1		45	25.3
k	176		3		2	2	10												1	10					1	13	42	23.9
l	176			1		3															1			1	7	4	17	9.7
m	176	15		2		3	3			1		11	6		1						2		1		3	1	49	27.9
n	176	1		4	1					1	18		5						2						2	6	40	22.7
ng	176	1		1		9				5	21	41									2				14		94	53.4
p	176	4			6	1	16		7										1	5					2		42	23.9
r	220	4		1		1				6												1			1	2	16	7.28
s	176				1	2													3	3			5			1	15	8.5
sh	176		11	1				1						1		2			1	2					1		20	11.4
th	176	4		6	19	2			10	1			1	1		3				10			2		7	2	68	38.6
t	220	1	1		2		13	1	16					3		1	1				1				5	18	63	28.6
v	176	33		5		8	1			1	1	1			1				2						9	7	69	39.2
w	176	1									4										3		2		1		11	6.25
y	176					6	1	1				1															9	5.1
z	176			5						1		2					9		3	2	11		1		3		37	21.1
	4400	97	24	47	34	82	49	6	51	21	52	63	10	12	7	21	7	28	30	57	3	11	12	117	65	955	23.6

If speakers in auditoriums would give more attention to the proper emphasis of the consonantal sounds, and especially to such consonantal endings as *ng*, *d*, *v*, *f* and *th*, the hearing of speech in auditoriums would be greatly facilitated. Public speakers, and instructors in oral expression, could make a substantial contribution to the problem of hearing in auditoriums if they would utilize this information in the training of the speaking voice.

In conclusion, it appears that we have expected altogether too much from our auditoriums. In a large auditorium, the loudness of a speaker's voice is at a critically low level, so that the slightest disturbance from noise, reverberation, or interfering reflections will result in unsatisfac-

tory hearing conditions. There is, therefore, an urgent need for increasing, in some way, the loudness level of the average speaker's voice. An improvement may be expected from proper voice culture, or from suitable reflecting surfaces near the speaker, but the principal improvement is to be expected from artificial amplification of speech, as by suitable public address systems. The improvement of apparatus for the reproduction and amplification of sound is progressing at a gratifying rate, and we may confidently anticipate that present and future developments in this art will make a most beneficial contribution to the problem of good hearing in large auditoriums. It is probable, when these developments are realized, that auditoriums which are to be used primarily for speaking will be designed to have a time of reverberation considerably lower than the currently accepted optimal time. In smaller auditoriums (smaller than about 400,000 cu. ft.), where no amplifying equipment is contemplated, the optimal time of reverberation can be determined from the data and curves presented in this paper. In addition, these data and curves make it possible to evaluate the acoustic merit of any auditorium for the hearing of speech, and thus they place this phase of architectural acoustics upon a more exact quantitative basis.

THE MECHANISM OF SPEECH

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"The quality, or 'timbre' of the human voice, I believe, is due in a very minor degree to the vocal cords, and in a much greater degree to the shape of the passages through which the vibrating column of air is passed."

This statement made by Alexander Graham Bell[1] in 1907 probably represents the general view of scientists today. Some would of course modify the terminology slightly, and say "*the quality is due to the volume of air in the cavity or cavities and size of the openings thereto.*" But the idea is the same.

We need hardly say that vowel quality differences are also generally ascribed to the function of this same volume of air in these cavities, acting: some would say as resonators accentuating certain harmonics in the glottal complex stimulating sound; while others maintain they function, to add independent cavity tones to the glottal concomitant which merge together as a vowel formant (if we may use the term coined by my old teacher Hermann).

For purposes of the present study, we need not enter into the dispute as to which of the cavity tone theories should be accepted. Whether we should take the fixed, or transient state theory. Whether we ought to follow the harmonic, or the formant theory. Whether the vocal cords stimulate these cavities by means of puffs, as do the lips in blowing a trumpet, or whether they actually vibrate, and hence, produce a series of harmonics in much the same way that vibrating strings or thick edged membranes would. All are alike in ascribing these vowel quality differences to a function of this total volume of air in the vocal cavities. And that is the only aspect of these theories we are concerned with, viz: *is this cavity influence—that is the total volume of air involved, and varying diameter of the openings thereto, responsible for all vowel and voice quality differences?*

In seeking the answer, the usual process has been to record the speech sound as it emits from the mouth, give the same a Fourier harmonic

[1] Bell, Alexander Graham. The Mechanism of Speech. Volta Bureau, Washington, D. C. (1907) p. 28.

analysis[2] and then imagine the physiology which would supposedly give the results noted. Thereupon the investigator makes the *ex parte* assumption that the imagined cavity air volume influences were responsible for all vowel and voice quality distinctions as manifest in the speech curve.

Even the most careful scientists have fallen into that unscientific way of dealing with the problem. This is especially true of physicists and others not at home in the physiological field. Curiously enough many such do not hesitate to state their own subjective physiological imaginings as fact or assume them to be such, when they would be the first to disclaim any attempt at scientific physiological experimental proof by way of verification. They resort in other words, to unsupported physiological surmises of an unscientific type they would be the first to condemn in their own field of physics. The all too common tendency has been to take the same attitude assumed by Willis a hundred years ago, who in introducing the report on his vowel experiments said he proposed:[3]

"*neglecting entirely the organs of speech*, to determine if possible, by experiments upon the usual acoustical instruments .... what conditions are essential to the production of these sounds."

It would seem we all would agree that actual experimental proof is always preferable to assumptions of that type built on analogy. Rather than guess at the physiological facts a scientist should demand some type of investigation which would record the physiology as efficiently as we record the physical manifestations of the voice curve. That has been the purpose of the present experiments.

Such a study demands a measurement of the complete cavity with its apertures as involved in the production of vowel and voice quality, with its dimensions along the line of widest opening in three planes: (1) From front to back. (2) Perpendicular or median. (3) At right angles to the latter. It follows of course that all of these should be made sim-

[2] Fourier, Théorie analytique de la chaleur, Paris (1822) Ch. III. Cf. Hermann, *Kurvenanalyse u. Fehlerrechnung.* Pflüger's Archiv.(1902) Vol. LXXXIX p. 600. Also articles in Vol. XLVII p. 47 and LXXXVI p. 92.

Scripture, E. W. Elements of Experimental Phonetics, Scribners N. Y. (1904) p. 561 ff. And Chap. XXVIII.

Rousselot, L'Abbé, Principes de Phonétique Exp. Last ed. appendix.

Miller, D. C. The Science of Musical Sounds, Macmillan, N. Y. Lecture IV for a complete treatment including modern machines for carrying out the process mechanically.

[3] Willis, Wilfrid. *On vowel sounds and on reed-organ pipes.* Trans. Camb. Phil. Soc. (1830) III 231.

ultaneously; and a record taken of the sound as it was at that instant, in order that the experimenter might listen back to the same and verify its quality when doubts arise in regard to unexpected manifestations in the cavity photographs. Since we seek the cause of differences it becomes the duty of the investigator to seek deviations from the assumed one and, actually stimulate the taking of other positions if possible in order to ascertain whether the one theoretically looked upon as the cause is actually responsible or merely an incidental manifestation. And in numerous other ways the experiment must be carefully controlled.

This we have earnestly attempted to do, and believe we have done. For the front buccal cavity lateral or horizontal measurements we use *palatograms*[4] which give an exact record of the cavity boundaries as far back as the *pillars of the fauces* (or *palatine arches*); and that is well up to the perpendicular line of the back throat cavity as manifest in vowels generally, so the horizontal measurements should not extend beyond that point. This record is accompanied by another of the lip spread from corner to corner. And an X-ray photograph is simultaneously made by our technique for showing the tongue, velum, and other soft surfaces within the hard bony head cavity, now finally perfected after years of exasperating experimentation.[5] These latter give for the first time, exact indication of the median dimensions of the cavity along the precise center line of widest opening from the front lips to the lower back glottal wedges (vocal cords). The lateral measurements for the pharynx and interior larynx are obtained by means of my laryngo-periskop, at intervals of 1 cm. from the velum down to a point from 3 to 5 cm. below the tip of the larynx, or in other words to the upper level of the glottal wedges or actuating source of the voice sounds.

[4] Rousselot, L'Abbé. Principes de Phonétique exp. for a description of the technique he devised. Uses a false palate of the type made by dentists, usually credited to Oackley Coles. We use a technique which marks the palate off into squares of 1 cm. each way, by punching a pin hole through at cm. squares; the impression of the teeth is kept, permitting the palate to be kept in a constant flat position along the same plane when laid down flat; a light placed thereunder permits of an exact tracing, of the contour and location of the dots whose altered proximity, thereupon shows the various slopes and indentations of the mouth roof. It follows that the point of widest lateral opening will always fall where the tongue touches the palate. After each X-ray exposure the palate is removed and this lateral record of tongue contact made at the same simultaneous instant is thereupon taken to be filed away with the record, as are the others, A representative collection will be found in Figs. 301 to 539 in my book "The Vowel" O.S.U. Press, Columbus, Ohio. (1928). And the ones made simultaneously with the X-rays herein for the American young lady Subject 291, on p. 269 Figs. 250 to 255, as the only ones which made any contact whatever.

[5] Technique described in "The Vowel" O.S.U. Press, Columbus, Ohio.

A few attempts at tongue position X-rays[6] have been made at various times in the past, but none of them with a view to recording the median plane at the point of widest opening in the cavity. And after all that is the absolute *sine qua non*, for the mere indication of where some indiscriminate part of the tongue is can throw no light on the question before us. They either follow the technique of Scheier, which uses no marker to delineate that median line, and hence shows only the outside curved-up edges of the tongue; or else they utilize only a partial length marker, or one that is allowed to wobble back and forth along

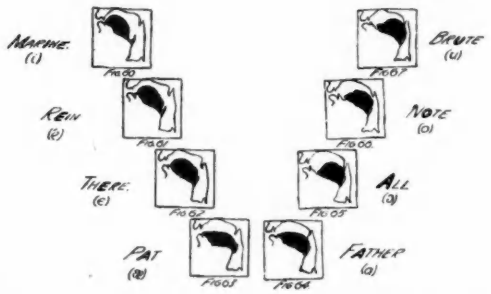

FIG. 1. *The traditional, imaginary, and highly fantastic, vowel cavity diagrams commonly used by physicists and other linquistic scientists to demonstrate the cavity tone theory. By Lloyd.*

the tongue in such a way as to do anything but delineate the point of widest opening—permissible for their purposes since they desired only to have some indication as to the general position of any part of the tongue. A large number of palatograms have been published, but never accompained by median computations; and some of isolated lip positions. No pharyngeal measurements have ever been made. So the present study is the first physiological investigation to be made of the cavity tone speech and voice theories. It includes some 3000 X-ray and accompanying measurements on over 400 subjects.

It is perhaps unnecessary to say that the median record shown by the X-ray is not only by far the most important, but also the most enlight-

[6] Other X-ray photographs. Meyer, E. A.. *Untersuchungen über Lautbildung.* Viëtor Festschrift Elwert, (1910) p. 172.

Scheier, M. Bedeutung d. Röntegnverf. Arch. Laryng. B. 22 H. 2 p. 175.

Barth u. Grunmach, Röntgenographische Beiträge z. Stimmphysiol. Arch. f. Laryngologie, Bd. 19, p. 396–407.

Eijkman, L. P. H. Tongue Position in the Pronunciation of Some Vowels. Tracings from X-rays, mostly conventionalized composites. Vol. June, 1914, Heft 3.

ening. That is because the tongue creates most of the changes in the size of the vocal tract, and its movements are largely in that median plane. The extent to which it can alter the vocal cavity through the outer curvature and lateral constriction of the tongue edges other than that indicated in the X-ray photograph (note the broken white lines) is much more limited. And the thing may be said of the dimension alterations which the pharyngeal muscles may create. The reader will undoubt-

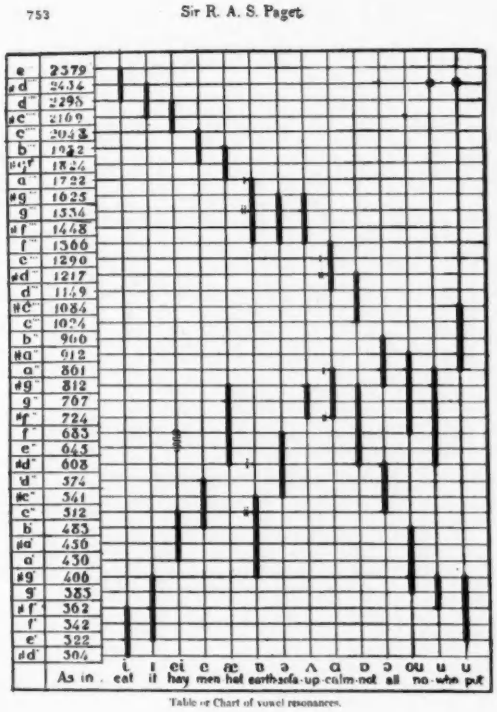

FIG. 2. *Paget's table of two tone characteristic cavity pitched for English vowels.*

edly be more interested in the X-ray record therefore, than in any of the others above mentioned. So let us first turn to the X-ray photographs of the young lady subject from the Mid-East U. S. as they are presented herewith.

Of course the easiest thing for me to do would be to attempt to interpret them in the light of accepted vowel theories; and seek to harmonize the tongue positions and cavity dimensions the X-rays show as an actuality, with the traditional ones which have grown out of the imagina-

tions of previous investigators, fantastic as the traditional ones are now proven to be. But if we are to be scientific in our examination of these experiments, we are called upon to divorce our thoughts from all preconceived notions. It behooves us to look at the facts whether they go contrary to our accepted and pet theories or not. And all may as well be warned in advance that they will undoubtedly do so in more than one respect. If I point out these iconoclastic manifestations to the relative exclusion of the conforming ones it is only because the latter will be most evident without any indication on my part. And I trust that procedure will not be misunderstood.

From the standpoint of the theories held by the physicist Fig. 3 giving what the X-ray shows to be the median tongue position and cavities for the vowel *i* (in *peep*) conforms best. Because as is evident, the vocal tract for this vowel presents a contour which reminds us a great deal of a relatively large bellied bottle (corresponding to the back throat) with a long narrow buccal neck turned at right angles after the order of a Spanish wine decanter. According to the cavity tone theory which would use the Sondhauss[7] mathematical formula this neck would produce the high pitch; and if we follow Helmholtz the back body would give the low pitched cavity tone found in harmonic analyses of the speech curve, or if we followed Lloyd and most modern physicists the total volume involved in the belly plus the neck would be responsible. Attention is called to the fact that the back throat or belly of the bottle is, however, even more regular in its shape than is the front buccal cavity or neck of the bottle; so it is therefore, quite as much like an organ pipe whose pitch is governed by its length, as is the front cavity. Helmholtz and others assume the back throat cavity to be closed by the glottal wedges but there has always been a question in my mind whether this could be held to be true where the glottis is open into the trachea as much of the time as it is closed. Besides it produces a cavity stimulus which should be loudest at that point. It is evident that the front buccal cavity is open at both ends, but what are we to say of the back pharyngeal?

The question as to whether this back throat cavity is to be considered as open or closed is of particular importance in a study of this problem as to the cause of vowel and voice quality, now that we note the regularity of this throat tube and the possibility that it might be responding

[7] Lloyd. *Speech Sounds: their nature and causation.* Phonetische Studien (Neuere Sprachen) III (1890) p. 275–278; IV (1890) p. 39; V. (1891) p. 125.

Sondhauss, earlier form Pogg. Annalen vol. LXXI p. 347–373.

American English. Young lady from Central Ohio. Do these tongue positions substantiate a one, two or three cavity-tone theory? What causes a supposed 2 resonator effect in the vowel of Fig. 6 and not in Fig. 8 or Fig. 11?

FIG. 3.
Vowel i (*ee in peep*).

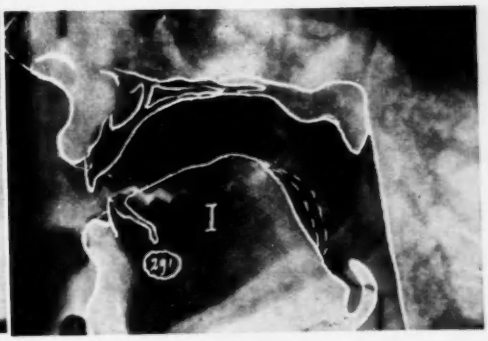

FIG. 4.
Vowel I (*-i- in pip*).

FIG. 5.
Vowel e (*a-e in pape*).

FIG. 6.
Vowel ɛ (*-e- in pep*).

FIG. 7.
Vowel æ (*-a- in pap*).

FIG. 8.
Vowel ɑ (*an in bah or balm*).

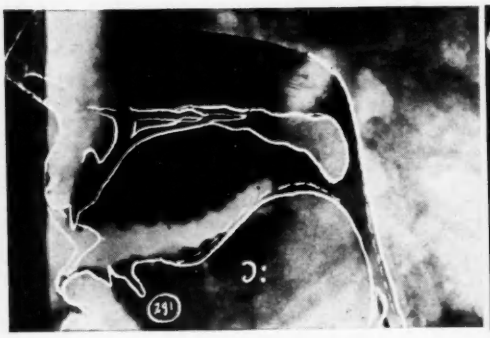

Fig. 9.
Vowel ɔ: (*aw in maw or bawl*).

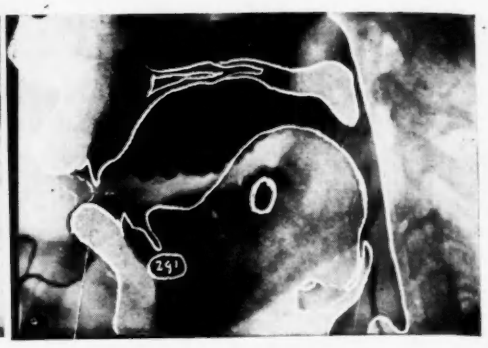

Fig. 10.
Vowel o (*o-e in pope*).

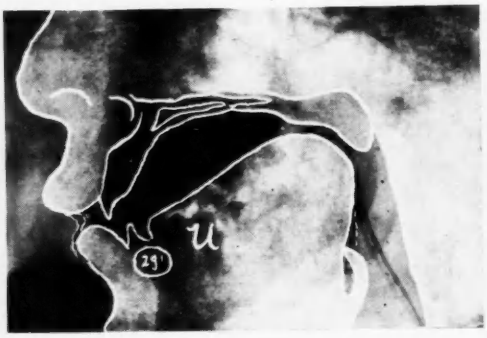

Fig. 11.
Vowel u (*oo in boob or rune*).

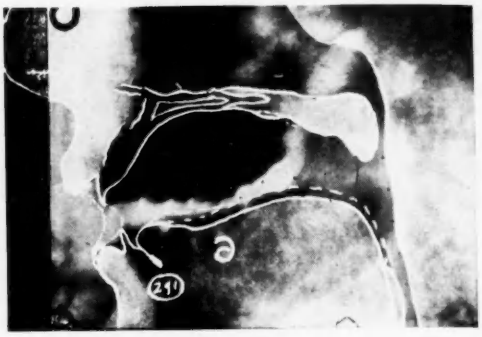

Fig. 12.
Vowel ə (*uh in idea, the or pup*).

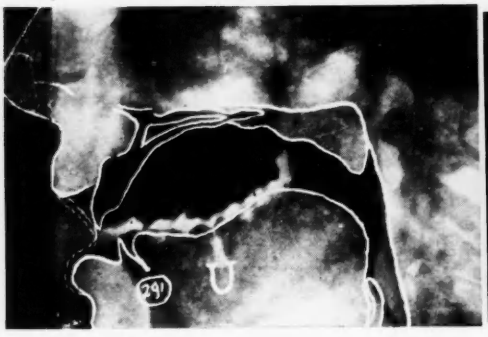

Fig. 13.
Vowel U (*oo in foot*).

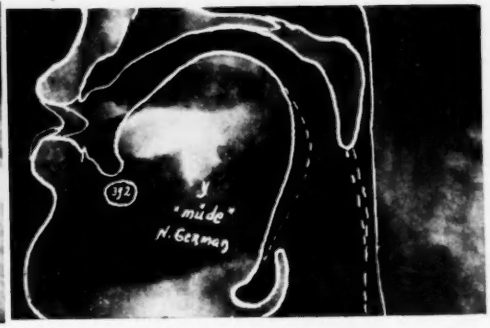

Fig. 14.
Vowel y (*ü in müde*) *with lip rounding.*

Note: But little difference between the tongue position for the vowels in *the* and *pup* was found.

to the same laws as the organ pipe. For we know that the open pipe yields the complete series of partials, odd and even alike, whereas the closed responds only to the odd, the 1st, 3rd, 5th, 9th, etc. And if we maintain all these back pharyngeal cavities are irregular in shape and hence respond in accordance with the laws governing the spherical resonators, it is very apparent we must say the same thing of the front cavity.

Another interesting point is raised when we note the apertures opening into these cavities, which have heretofore usually been given little consideration. Observe Fig. 9 for example which shows the vowel ɔ (*aw* as in *bawl*) with a decided aperture just above the pharyngeal cavity opening into the nasal passages. What could we expect to be the resultant effect on the cavity tone? I note Helmholtz said:[8]

> "The opening of side holes is by no means a complete substitute for shortening the tube . . . . the upper partials of compound tones produced by a tube limited by an open side hole, must certainly be in general materially deficient in harmonic purity, and this will also have a marked influence on their resonance."

Aside from the mere aperture aspect in its effect on the buccal and pharyngeal cavity, one needs also to take into consideration the cavity tone effect coming from the injection of these two additional or a dually divided nasal tube. Being open at both ends one naturally expects the effect of a tube which would respond to all partials; yet nasal quality according to Auerbach[9] is heard where the fundamental is sounded with odd numbered partials only. Does the opening of the nasal passages make the vowel "nasal"? How? Why? If not why does any speech sound become nasal?

It is of interest to note that the *o* (*oh*), Fig. 10 and ə (uh) Fig. 12 are the only other vowels in the series produced by this subject which manifest such an opening into the nasal passages. And that nasal aperture for these latter two is much smaller than for the first. But thisobservation by no means holds true for all individuals.

Fig. 5 giving the X-ray photograph for the vowel *e* (as in *fame*) is another of those which could conceivably be interpreted as being clearly divided into two parts of a bottle with its belly and the neck thereto. So it readily pairs in this respect with the *i* (*peep*) of Fig. 3. Besides it is wider and hence contains a larger volume of air. Considered

[8] Helmholtz. English Edition, Sensations of Tone, (1912) Longmans & Co. N. Y. p. 102.

[9] Auerbach, F. Akustik 2nd ed. (1909) being second volume in Winkelmann's Handbuch der Physik, Leipzig, p. 263.

from that standpoint this neck for *e* (*fame*) should therefore give a lower pitch than that for *i* (*peep*).

But the application of the bottle equation of Sondhaus as made by Lloyd[7] and scientists generally, would postulate a progressively longer neck or front buccal tube, as one passed through the vowel series *i*, (*ee*); *I* (in *pip*); *e* in (*fame*); ɛ (*eh*); *æ* (in *pap*). The student of these X-ray photographs of the tongue position for the vowels will soon discover this postulate is not supported; for the length of the front buccal tube is essentially the same for all the vowels in the above series if one considers the hump as the termination thereof, since that regularly takes place right under the juncture of the hard and soft palates.

In other words the flask, or bottle and neck type cavity postulated by the usually applied Sondhaus mathematical equation is not verified by the X-ray photograph. We now see that the actual tongue position and resulting cavity is radically unlike that Helmholtz implied[7] and Lloyd drew. Under his scheme (See Fig. 1) the tongue arches up against the alveolar ridge and begins to drop off abruptly, postulating, thereby, a short cavity which extends to the alveolar ridge, and an extremely large one from there on back. Actually the arching really takes place, as you see, all the way back to the soft palate. In other words, we have a very long neck here and a relatively large cavity behind. The actual length of the two tubes (front and back) however, is essentially the same, as you will note. And it would probably be wise, therefore, to consider them as functioning like irregular, or spherical resonators, rather than as pipes— which most investigators since Sondhaus have been doing for front vowels.

On the other hand, I want to call your attention to the fact, that many of the vowels are regularly manifest in different subjects with cavities which can hardly be considered as dual. And, curiously enough, these are found among the front vowels, which have been considered by all investigators as having two characteristic "resonances." Fig. 4 is typical. The cavity is essentially all one, from the exit of the sound at the epiglottis right out to the teeth, and it would be only with a far stretch of the imagination that one could see how two cavities could be read into such a photograph, yet this Fig. 4 is for the vowel *I* (in "pip"). This same single cavity is manifest for the vowel *E* (in "pep"), Fig. 6. You will note that Fig. 2 shows they should have two characteristic, cavity tone, "resonances." Fig. 7 likewise falls in the same group—showing that one long cavity from the epiglottis straightforward to the lips, unless, perchance, you would like to ascribe the

high pitch to the cavity formed between the tip of the epiglottis and vocal cords—in other words to the laryngeal cavity. And if you do that, you must of necessity, ascribe a very high pitched characteristic frequency to the vowel "aw" ɔ (as in "*bawl*" or in "*awe*") in Fig. 9. For this cavity running from the velum straight down to the larynx certainly would, of itself produce a pitch quite as high as that for the front cavity of the vowel "ee" (*i* as in "*peep*"), of Fig. 3. Evidently it is quite as narrow and about the same length. The essential difference lies in the fact that the narrow "aw" (ɔ) cavity is lined with back throat soft tissues, and the "ee" (*i*) cavity is created by arching against the front mouth hard surfaces. The "ah" (*a*) Fig. 8, would likewise fall in the same catagory, and yet, Miller found but one characteristic frequency for both of these latter vowels. Of course both these latter cavities are megaphone-like. But why the elongation of the narrow back throat tube for the "aw" (ɔ)? It occurs as a characteristic regularity.

That brings me to a consideration of this question of surfaces. I am frankly surprised that none of those who have studied the vowel should have ascribed to surfaces and their influence any of the qualities manifest in vowels. Professor Miller, the other night, called our attention to the possible influence of surfaces in changing the quality of flutes. In his science of musical sounds[10] he reports his experiments with an organ pipe having double-walls between which he could pour water, and noted that during the filling the "quality changed, conspicuously thirty or forty times," even though the dimensions of the "resonance" cavity itself were kept constant.

If surfaces create quality differences between musical instruments, why should they not do likewise in speech and voice? Why should they not be involved in creating vowel quality differences?

Is it not possible that this is the physiological reason for the creation of that long soft surfaced chimney through which the sound is forced to escape for the vowel ɔ ("*aw*" as in "*bawl*"), and for the difference between this vowel and that of the vowel *a* ("*ah*")? These are soft surfaces in that back throat neighborhood and soft surfaces must inevitably either muffle the sound, or mellow its quality, especially where the tube is so narrow in relationship to the total area. If they were hard, the opposite effect might be expected, giving a metallic tonal quality of sound somewhat analogous to that produced by extremely narrow organ pipes.

[10] Miller, D. C. Science of Musical Sounds, Macmillan, N. Y. (1922) p. 181.

One might, therefore, expect by reason of the surface involvement, a distinct difference between the vowel ɔ ("*aw*" as in "*bawl*"), in Fig. 9, and the *i* ("*ee*" as in "*peep*"), Fig. 3. Both of them show a long narrow tube joining onto a relatively large-bellied cavity. For the first, the long neck is created between soft surfaces, and for the last, against hard surfaces. Otherwise, they would both be narrow and long, and if manifest in organ pipes would be expected to produce high-pitched characteristic frequencies.

Of course, there is a great deal more evidence which tends to justify us in this theory we lay down in regard to hard and soft surfaced influences on vowel and voice quality. I need not call your attention to the fact that soft piano hammers produce a much "mellower" tone than hard metal or wooden hammers. That fact has long been known, and the physicist has no difficulty in explaining it. We say that, the soft surface wipes out the extremely high-pitched "metallic" paritals, by reason of its tendency to cling to the string, and hence "mellows" the tone. We know that the same thing happens when we pass the sound through a tube line with soft surfaces. The sound we hear coming out at the other end is quite radically different from the one heard before the soft surfaces filtered out certain of the frequencies. As it emits it is much mellower in quality—much softer.

As we know, certain of the vowels are of themselves, much mellower and even deader than others. That difference is heard as between æ ("*pat*") and ɔ ("*aw*" in "*bawl*"); for the latter is much mellower than the former, which is quite metallic in quality. On the other hand, the *u* (*oo*) is quite dead, even in comparison with ɔ ("*aw*" in "*bawl*"). Whereas, the *i* ("*ee*" in "*peep*") is distinctly the most metallic of all the vowels, and is the one which arches most decidedly up against the hard palate surfaces—surfaces in other words, which might accentuate high partials.

Now let us return to the cavity manifest in Fig. 4. Of course there can be no question as to the presence of the dual cavity tone in the speech curve for the *I* (as in "*pip*") which Crandall[15] for example analyzed, or in those which others found—though they disagreed widely as to what the characteristic pitch should be. The question is whether it would have been found in the recorded speech curve whose tongue position shown by the X-ray photograph in Fig. 4 we are now considering. In this case the actual position taken fails to show the two cavities required to produce the two cavity tones. What conclusion are we to

draw? Are we to say the two cavities are the cause and therefore necessary in order to produce the vowel quality difference which distinguishes this vowel from others in the series? Or were they merely an incidental manifestation in the curves Crandall and others have analyzed? Or are they but one of several other heretofore overlooked or disregarded or minimized causal factors?

It is granted that the average speaker may take such tongue positions as would create the speech curve. But we seek the *cause of vowel and voice quality differences*. And if it is possible to produce the distinction without this two cavity effect resulting from the tongue arch against the hard palate with a consequent narrow neck attached to a distended throat cavity or bottle-like belly, we are forced to draw one of three conclusions either:

1st. *My experiments or the ones of those who analyzed speech curves for this vowel are defective and misrepresent the facts.* I take it, this conclusion must be definitely ruled out as untenable. In my own mind there can be no doubt as to the entire validity of the experiments performed by Miller,[10] Crandall,[11] and others. And to give the reader more assurance in regard to my own, I am citing below an experiment all may perform themselves.

2nd. *Some compensations might be made by the vocal mechanism.* That would mean in this case that another cavity similar to the front buccal would have to be created someplace. And the X-ray photograph in question, Fig. 4, definitely disproves that conclusion. As noted above, the laryngeal cavity is the only one a stretch of the imagination could point to. And its presence in the positions taken by other vowels where the high pitched cavity tone is ruled out, would set that assumption aside. Cf. *a* (*ah* as in *calm*). with Fig. 8 and see cavity tones shown by Paget in Fig. 2.

3rd. *The two cavity tone manifestation found in the recorded speech curve was but an incidental factor among others of more importance.* And in that case the real cause of the vowel quality distinction which the ear hears must be sought elsewhere.

[11] Crandall, Irving B. *The Sounds of Speech.* Bell Technical Journal, Vol. IV, p. 612 No. XII.

German series of vowels. The general contours are the same as those of the American and Spanish subjects. The differences are quite of the type between those of the same language. The tongue position changes resulting from the unrounded lips is striking. Compare Figs. 14 and 15; 16 and 17. How do we explain the differences between Figs. 11, 29, 34?

FIG. 15.
Vowel y (*ü in müde*) *without lip rounding.*

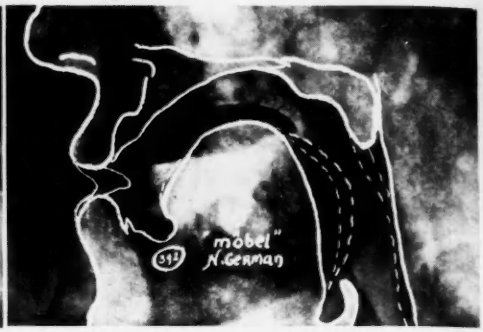

FIG. 16.
Vowel ø (*ö in möbel*) *with lip rounding.*

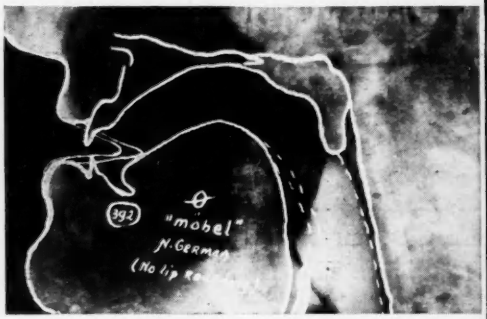

FIG. 17.
Vowel ø (*ö in möbel*) *without lip rounding.*

FIG. 18.
Vowel Y (*ü in mütter*) *with lip rounding.*

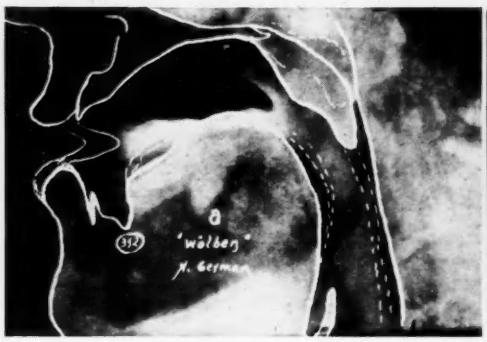

FIG. 19.
Vowel œ (*ö in wölben*) *with lip rounding.*

FIG. 20.
Vowel i (*ie in biene*).

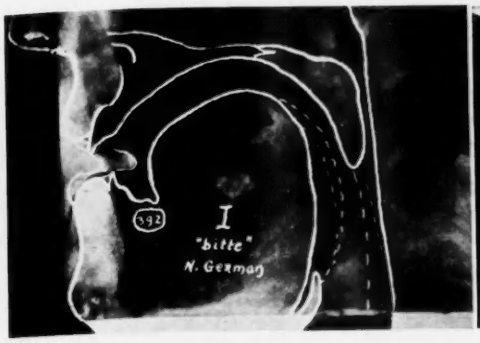

FIG. 21.
Vowel I (*i in bitte*).

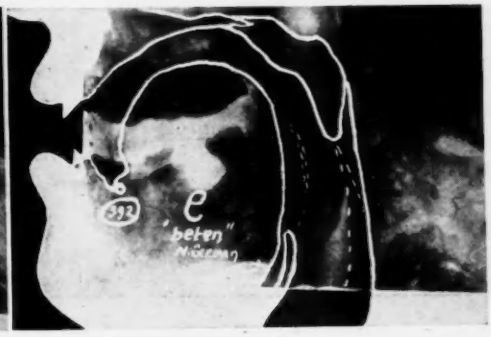

FIG. 22.
Vowel e (*e in bete*).

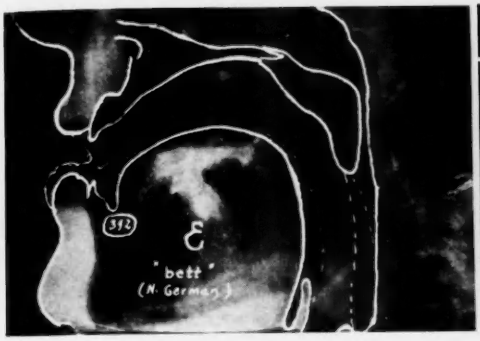

FIG. 23.
Vowel ɛ (*e in bett*).

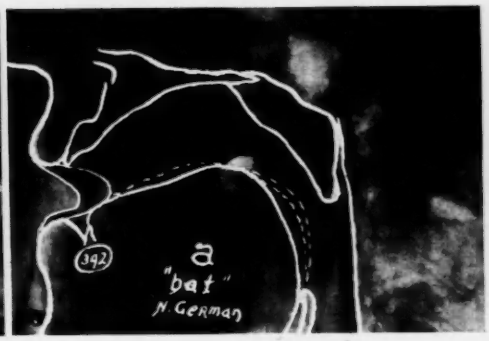

FIG. 24.
Vowel a (*a in bat*).

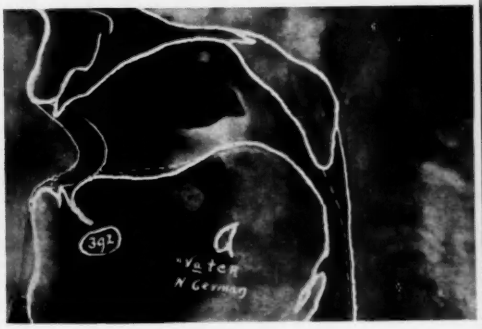

FIG. 25.
Vowel ɑ (*a in vater*).

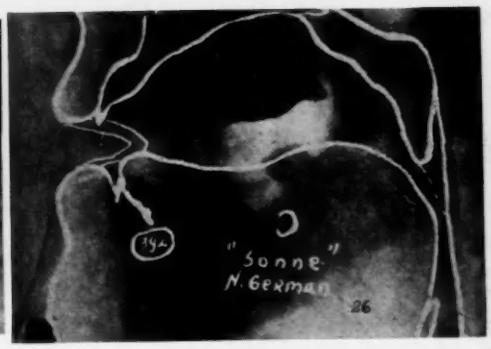

FIG. 26.
Vowel ɔ (*o in sonne*).

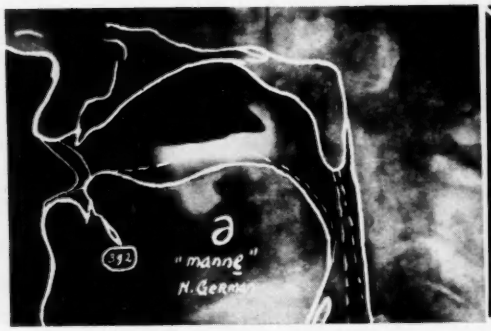

FIG. 27.
Vowel ə (*final e in maune*).

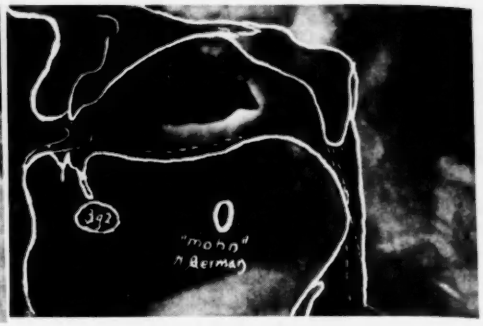

FIG. 28.
Vowel o (*o in mohn*).

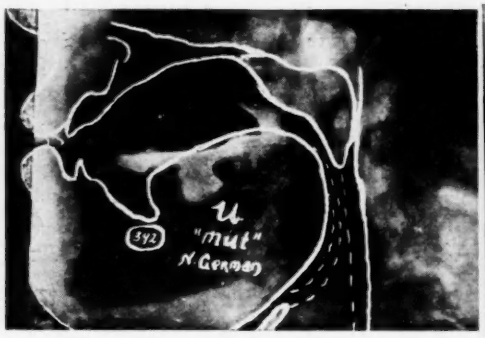

FIG. 29.
Vowel u (*u in mut*).

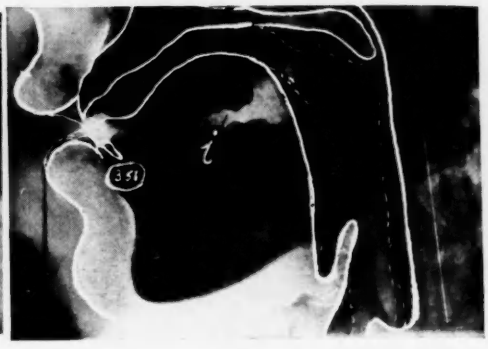

FIG. 30.
Vowel i (*Spanish i in si*).

FIG. 31.
Vowel e (*Spanish e in sé*).

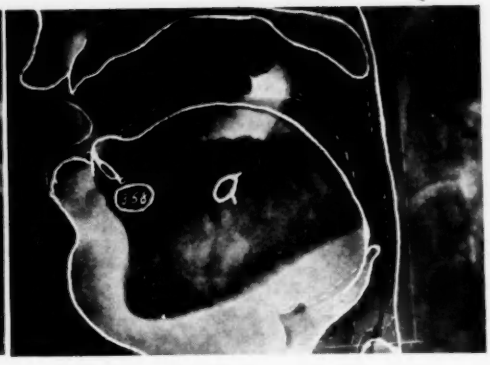

FIG. 32.
Vowel a (*Spanish a in la*).

Though it is admitted the factor in question might be a required one among others, some of which might be more important than this one.

A simple experiment will prove to anyone that it is not necessary to arch the tongue up against the hard palate, thus forming a narrow, bottle neck-like, tube, in order to articulate a perfectly good *I* (as in "*pip*"). A little practice will shortly demonstrate to everybody's satisfaction that it is possible for you to press the front part of the tongue down flat in the mouth with your index finger, and yet articulate an unmistakable "*pip*" and while maintaining the same position one can make the transition to "*pep*" without difficulty. All of which

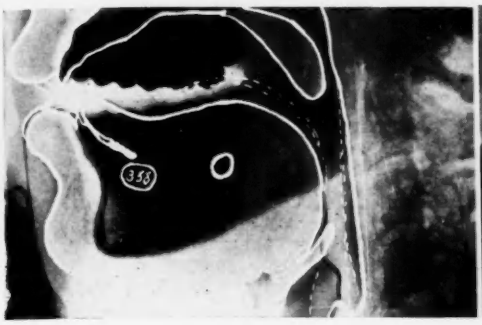

FIG. 33.
Vowel o (*Spanish o in no*).

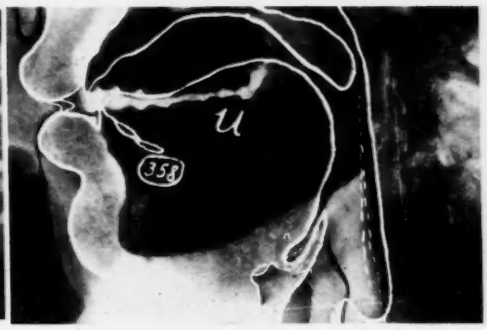

FIG. 34.
Vowel u (*Spanish u in su*).

goes to prove conclusively to any unbiased mind that the traditional modification in the front buccal cavity, and tongue position is not necessary in order to produce this vowel quality difference. And at the same time this simple experiment would indicate that other factors besides the cavity tones in question, must be intervening to create differences between vowels.

You may prove that same fact with other vowels. Try it. Just open your mouths and say "ah" (*a*). Most of you will say that your tongue lies flat in the mouth. When the vowel is pronounced in an isolated manner that way it commonly does. In rapid speech, however, it takes all kinds of positions. You will note that this Fig. 8, shows a decided hump in the back. That is hardly what we would have expected in terms of the older physiological theories. But, if you will now all of you raise the back part of the tongue up towards the velum you will note that you can push it up so high that the back part of the

tongue touches the wisdom teeth and extends well up beyond that, touches the soft palate and occupies a front buccal position fully as high as that of the vowel "oh" (*o*), shown in Fig. 12. That, of course, is a disturbing observation, because it is contrary to every physiological theory thus far postulated, and certainly does much to completely explode the old vowel triangle theory and Lloyd's physics which he tacked on to the same, postulating that quality difference manifested between the "ah" (*a*), and "oh" (*o*) as due to just such a rise in the back part of the tongue and its division of the vocal tract into varying types of cavities.

On the other hand if you will now pronounce with me the vowel "ah" (*a*) again, and press the front part of the tongue up forcibly against the teeth and the roof of the hard palate, you will note that it is perfectly possible to get a clear quality vowel and take a front tongue position almost identical with that manifest in Fig. 3, for the vowel "ee" (*i*). As a matter of fact you may also take a traditional consonant position as for L with a point lingual occlusion and a lateral opening which has heretofore been thought of as being the cause of this L-quality, and still produce a perfectly good "ah" (*a*). What are we to say to that?

Then again it is possible to narrow the lips until the opening is hardly as large as your pencil and still produce a good "ah" (*a*). Or, you may jerk the lips wide apart and make an opening as large as is in any manner of means possible and still get the vowel "ah" (*a*). Will you narrow your lips thus and pronounce the vowel for me? Now open them wide and repeat the process. You note that both of these positions create perfectly normal quality in the "ah" (*a*). Some will be prone to assume that compensations take place to keep the cavity the same size, or to retain the same cavity tone pitch by making corresponding changes in the apertures at other points. I thought so too, until I sought experimental proof by means of X-ray photographs taken at the time. Suffice it to say that it was not forthcoming, for other parts of the tongue position did not change materially, or enough to account for the radical shift in lip position.

That brings me to a consideration of the tongue position shown in Fig. 13, for the vowel "oo" (*u*). Now I should like to reiterate at this point, that simultaneous phonograph records were made of each vowel taken at the instant of all these X-ray photograph exposures. So, there can be no question as to the quality produced during the tongue position in question. That synchronized, dual phonograph and cavity

dimension record was provided in all of the 3,000 experiments on over 400 subjects made covering this investigation.

I may say that the position taken in Fig. 13 for "oo" (*u*) was one deliberately assumed by the subject in question. She was asked to pronounce the vowel "oo" (*u*) in "poop" while pressing the tongue up tightly against the roof of the mouth. Nobody who listens to the record would mistake the quality. Yet it will be noticed that the front mouth cavity is, if anything, considerably smaller than that manifest for either the vowel *I* (in "*pip*") of Fig. 4, or even the vowel *e* (in "*pape*") of Fig. 5, though of course its openings are smaller. We likewise note that the total capacity of the front cavity is considerably less for this vowel "oo" (*u*) than it is for the vowel "ee" (*i*), in "peep" of Fig. 3. And I may say that the narrowed lip or velar opening will not suffice to account for the failure of this cavity to produce the same characteristic high pitched frequency manifest in the vowel "ee" (*i*). For, since that front cavity is tube-like in formation the best that this narrowed opening could do would be to reduce the front cavity tone pitch by one half, which would bring it down in the neighbrohood of 1500 cycles per second. Whereas, Paget finds, (see Fig. 2) a characteristic high frequency of any place between 608 and 861 cycles per second for the vowel "oo" (*u*), and around 2500 cycles for the vowel "ee" (*i*). Of course, the coupled back cavity will lower the pitch of the front one; and the lower the pitch of the back cavity the more it would lower that of the front. That is important for it will be noticed that the back cavity of Fig. 3 is very considerably larger than that of Fig. 13, and enough, so as to fully compensate for the decreased velar opening manifest in the latter.

Will you repeat that experiment and say "oo" (*u*) while pressing the front part of the tongue up forcibly against the teeth and hard palate? Now, if while you hold that tongue position you will pass quickly to the vowel "ee" (*i*) by jerking the lips back and spreading them apart without moving the tongue in the least, you will notice that it is possible to make the transition required for "we" (*ui*) without calling for any change in front tongue position whatever. My moving picture X-ray photographs show that this is what we regularly do when we pronounce the word "we."

On the other hand, if you will pronounce now for me an "ee" (*i*) with your lips puckered as if you were pronouncing the vowel "oo" (*u*), you will note that it is not at all impossible to get a good quality "ee" (*i*) by such means. This is disturbing because the customary

physiological theory ascribes this combined position to the characteristics for the rounded *y* as in the French "tu" and German "müde"—that is the vowel the latter call the "umlaut *ü*" (*y*). Yet you just proved to yourself that you can say "peep" with a rounded lip position for the vowel which leaves an opening not as large as your pencil between your puckered lips.

I want you to note now, that there are Germans who are able to pronounce an "umlaut *ü*" (*y*) without any puckering of the lips whatever. That is shown in Fig. 14. And, I should like for you to compare that with the Fig. 15 showing the puckered lip position. Then pass to the German "oo" (*u*) of Fig. 16, and the German "ee" (*i*) of Fig. 17. Can you account for all these varying vowel quality differences by using the customary total air volume resonator cavity tone theories alone? I am frank to confess that I have been unable to do so.

But some of these vowel differences are manifestly due to cavity tone influences and this vowel "oo" (*u*) is a good evidence of that fact. The cavity tone influence is, of course, quite unlike that which has heretofore been imagined. Practically nobody has paid any attention to the back pharyngeal cavity, but you will note what a distinct enlargement of that cavity takes place in the production of this German "oo" (*u*). You will observe what a radical enlargement of that cavity is created in the lower pharyngeal area right where the sound emits from the larynx.

This enlargement is in part created, not so much by the function of the tongue muscles, as by those which pull on the hyoid bone. That distention is regularly noted, not only for the vowel "oo" (*u*) (see Fig. 16) but also, for the vowel "ee" (*i*). (See Fig. 3 and Fig. 17).

You may all sense that pull on the mylo-hyoid and the genio-hyoid muscles if you will just press upward with your thumb tightly right under the tip of the chin. Now say first "ah" (*a*) in which the muscles are relaxed, and then "ee" (*i*). If you pronounce the two vowels as a successive alternating series, separating each element in the pair, repeating slowly thus "ah-ee ah-ee ah-ee" you will notice a distinct tension of the muscles under the chin on the "ee" (*i*). This is necessary in order that the back pharyngeal cavity may be decidedly distended.

Of course it is evident that this distention creates a decided resonator for several of the vowels such as "oo" (*u*). Besides that resonator gets progressively smaller as we pass from "oo" (*u*) to "oh" (*o*); then to "aw" (ɔ)—in which the back pharyngeal resonator is usually almost

completely eliminated; and on to "ah" (*a*), where the resonator is prevented by bringing the tip of the epiglottis right back against the pharyngeal wall, raising the larynx and flaring the mouth opening from that point upward and outward in very much the same manner as one does with a megaphone.

On the other hand, there must be some other distinct reason for the varied shape apart from the distention of this same back cavity in the front vowels. The very manner in which the tongue changes its slope from one of these vowels to the other without at times actually altering the capacity of the cavity or the fundamental dimensions argues against a cavity involvement or a resonator effect. But that variation in slope cannot be without significance, for it is quite characteristic. If not, then why is it that the tongue changes its shape without regard to cavity volume and merely slopes progressively more and more back in the direction of the lower pharynx as we pass through the series: *i* (peep); *e* (pape); *I* (pip); *E* (pep); *æ* (pap)? Study Figs. 4, 6, 7, and 3, 5, 8, comparatively. That gradual relaxation of the *genio-hyoid* muscle, with a resulting backward slope of the tongue in the neighborhood of the epiglottis is a regular and characteristic manifestation for these vowels. And the movement is generally progressive as you will see. In other words, the shape the surfaces lining the cavity assume, the tension manifest therein, and the manner they muffle the sound or accentuate the high pitched metallic partials by narrowing a soft surfaced tube on the one hand, or a hard surfaced one on the other, would seem to have some importance in changing the quality.

May we not therefore conclude that the total volume of air in the vocal cavities functioning as a resonator, or resonators cannot be the only factor involved, in the creation of either vowel or voice quality differences?

At least that conclusion would seem to be justified if we can vary the capacity of the cavity at will without making compensations in the apertures, and yet keep the quality of the vowel clearly recognizable. This we did above for the vowel *I* (*pip*); *E* (pep); *a* (pop); *u* (poop); and we might extend the list well beyond that number.

But we have much better evidence than this just cited. If the vowel differences were traceable solely to the total volume of air involved in the cavities above the vocal cords, one might expect to be able to duplicate those cavities so as to include exactly that amount of air and be able to reproduce with unmistakable fidelity, the quality manifest in the normal human original. Does that not seem to be a resonable assumption?

As already indicated we have succeeded in getting simultaneous measurements[13] of the vocal tract cavity (or cavities if you prefer to speak of them as such) in three planes. That makes it possible to reconstruct the cavity with the total volume of air involved just as the original human subject used it to produce the given vowel in question. This reconstruction we proceeded to carry out in clay for a whole series of vowels. You will find the report of results as well as computations for the complete series of vowels on a large number of subjects published in my book on "The Vowel" by the Ohio State University Press. So there is no need to cite the same here. Suffice it to say that the results gave an imitation of the vowel no better than you heard during these sessions produced by the artificial larynx. And I believe you will all agree that the vowels thus produced bear such slight resemblance to the human production, that nobody would be deceived.

Our results were no better than the innumberable synthetic, and other artificial machine reproductions such as those of Helmholtz, Koenig, Rousselot, Preece-Stroh, Miller, Stewart, Paget, and the host of others, some of which many of you have heard. I believe most of you will admit nobody would be deceived into thinking them human. They are at best but bare imitations. And in all frankness I have to admit our results were of the same type. Alexander Ellis the famous phonetician who translated and annotated the work of Helmholtz so well as to make our English edition infinitely superior to the original, was scientist enough to admit the artificial deficiency. And none of our recent ones are any better, in my opinion, and it has been my privilege to hear most of them. He said:[12]

"The synthesis of Prof. Helmholtz and that of Messrs. Preece and Stroh . . . . prove distinctly that difference in the quality of tone, taking only *harmonic partials, is the foundation of vowels.* Both, however, also *prove that there is much more yet to be learned before we can satisfactorily imitate spoken vowels* . . . . . Each speaker has his personal quality of 'voice'; . . . . so that there are really millions of different qualities of tone all recognised generically as the same vowel. And yet in the artificial vowels just considered *I could not recognise any exact form of human vowel with which I was acquainted*, although I have made speech sounds an especial study for more than forty years."

Who could say more of those he has heard. Of course the creator usually insists on the perfection of his reproduction. It is a very easy matter for any of us to drift into a self-hypnotization,

[12] Ellis, Alexander J. See his Appendix at the close of his translation and annotation of Helmholtz, Sensations of Tone, Longmans & Co. N. Y. (1912) p. 543.

when the hearing is concerned. We hear what we are listening for. So every such reproduction must of itself be carried out as a carefully controlled scientific experiment, with a large number of observers comparing the original with the synthetic reproduction and recording their judgements. And that scientific test of artificial efficiency[13] in reproduction has rarely been made. In "Speech and Voice" I publish such a controlled test[14] on the results produced by Sir Richard Paget.[15] It bears out what I have just said. Yet I must say his artificial vowels are by far the best I have heard. And by the way I must say I admire my friend Paget more than I can indicate in words, for submitting his results to such a test, for many others making such claims have begged the question, and put it off in one way or another.

While I am discussing Paget's work I want to cite two other proofs. First, Paget can cup his hands into a single cavity with a reed between the thumbs, and make them say as cleverly as any artificial reproduction I ever heard:

"Laila I love you" and "Hello London are you there"
(laila ai ləv ju) (ɛlo lɔndɔn a ju ðɛ˞)

What conclusion are we to draw? He uses but a single cavity. Does that result indicate a two cavity resonance as necessary to create the quality distinction in any of those vowels? Or are we to say the ear of the observer is deceived slight-of-hand-like into hearing something which was not reproduced by this mechanical process? Either would disturb him. And should also all those who hold the cavity tone theories are definitely proved as all-sufficient explanations of the vowel.

Second, when I asked Sir Richard to raise the pitch of his reed, corresponding to the voice or glottal tone, during the production of his two-tone resonator vowel ɛ (*eh*) the quality changed without any alteration in the capacity of the resonator, to that of the vowel *i* (peep), or at best something like that of *I* (pip). Why?

Perhaps I may now return to our own experiments in the reconstruction of the vowel cavity. It is not pleasant to have to acknowledge them to be but inefficient imitations of the human original. But in all scientific honesty I must make the admission. For we submitted our results to the test above mentioned. First we thought the failure might be due to the fact that the cavity tone pitches were slightly

[13] Russell, G. Oscar. The Vowel. Ohio State University Press. (1928).
[14] Ibid. Speech and Voice, Macmillan, N. Y. (In press).
[15] Paget, Sir Richard. *Artificial Vowels.* Proc. Roy. Soc. Vol. 102. (1923) p. 755–762.

inaccurate. As I said we made the heads of clay. And it is a well known fact that soft walls for a cavity lower its pitch. That fact has not been generally noted by cavity tone apologists in their treatment of vowel physiology. They talk of the total volume of air and the apertures. But the surfaces also have an effect in changing the mere pitch of the cavity. And surfaces of either human bone or flesh are not like those of clay.

Our next step led us therefore to make adjustments in the cavities such as to make them speak to tones within the pitch range cited by Paget. See Fig. 2. I have such a head here. It is stimulated with a freely vibrating organ reed pitched at 256 d.v.—that used by this particular subject, during the production of the X-ray experiment. Let me stimulate it. How many hear "ee" (*i*)?—(Scattering show of hands) How many "eh" (ɛ)? (Large number of hands) How many "ah" (*a*)? (About the same number of hands) How many "aw" (ɔ)? (About the same) How many "oo" (*u*)? (Fewer hands). How many failed to raise your hands? (Forest of hands up, and a laugh) Well the vowel this head is supposed to represent, and to which its cavities are tuned in accordance with the table of Paget, is the vowel "ah" (*a*). (Laughter). It is, frankly speaking, our best, and the one I was in hopes most of you would agree upon (Laughter).

Naturally it next devolved upon us to ascertain the reason for this failure of the reproduced cavities to reproduce the vowel. Why should that be true?

The reed does not function like the vocal cords is the first idea which strikes us. You note that the quality of this reed in the cavity is "mellow" and pleasing like that of a well modulated singing or speaking voice. In that respect it is probably as good as that used in the artificial larynx demonstrated the other day. We tried 160 different types of reeds and other actuators, including as many imitations of the human larynx as we could get. The one I have used gave the best results.

But I still think the point made is worthy of consideration. I wish I could have you all up here to watch the marvelous play of the actual vocal cords, as they function during the production of vowel and voice quality alterations. I have here several of the pieces of apparatus I have designed in order to carry out such studies. In my "The Vowel" I have reported some of the extensive observations which have resulted from those studies.[13] And in my forthcoming[14] "Speech and Voice" to be published about Christmas by Macmillan, I have gone into the

question in detail. In general I may say they leave no question in my mind longer in regard to the validity of the surmise expressed by Ellis who said in the appendix to his translation of Helmholtz:[12]

"It appears to me that the *mode of vibration of the vocal chords is a most important element of vowel character*,and that the resulting effect is modified by the resonance in the ventricles of Morgagni, in the cartilagenous larynx more or less covered by the epiglottis (acting, possibly like, the cup mouthpiece of brass instruments) in the pharynx and between the pillars of the velum. . . . . By the original mode of vibration of the vocal chords for spoken vowels many inharmonic proper tones are probably produced. . . . ."

And why should that conclusion not be true? Surely nobody can deny the fact that the actuating source of sound in all kinds of musical instruments is of itself capable of making innumerable changes in quality differences through alteration of its own concomitant of partials and sometimes inharmonic components. The harpist changes the quality by altering the point where the string is plucked. The flautist by varying the kind of blowing. The cornetist not only by the use of the deadening mute in the bell, but by varying the type of lip stimulation, etc. If we stimulate the piano string by striking with the soft felt hammer provided, the tone is "mellow" whereas if we flip the same string with the finger nail, or strike it with either a metal or wooden hammer, it becomes "metallic." If I shift this reed used in the clay head the least bit so it strikes the metal edge as it vibrates back and forth the quality becomes "metallic" and even disagreeable. Is there any question but what the vibrating source may itself alter quality in the voice, and perhaps also in the vowel? Personally I have no doubt about it, and neither would you if you could see the cords function by means of one of these pieces of apparatus.

This small one, my *laryngo-periskop*, serves us best. You note its diameter is not much greater than that of a match. And it goes in around the tongue in such a way as to permit of normal speech and song, even the articulation of *S*, *T*, *D*, *K*, and the other consonants without impediment. And the speech, sounds natural as you hear (Demonstrated with the apparatus in the mouth). It also makes it possible for the subject himself and one observer to make the examination at the same time. This other is my *peri-laryngoskop*;[16] this my *optical laryngoscope*;[17] this my *glottal endoskop*;[17] and this my *fono-faryngoskop*[16] with the attachment for multiple observation, so that the subject himself, the examiner and a whole roomful of observers

[16] Manufactured by Electro-Surgical Instrument Co. Rochester, N. Y.

[17] Manufactured by American Cystoscope Co. Bronx, N. Y.

may see what is taking place. The first and the last have given me my best results, from a scientific standpoint. But the second is best for subjects with a tendency to gag; and the third or fourth for the average phonetician, or for the doctor making ordinary medical examinations.

You will note that the first uses the principle employed by John Dutton Wright (who is present) in his *auto-inspection laryngoscope*, and copied into the *auto phonoscope* by a German firm which some of you have heard of as the only piece of apparatus for making such examinations. The results I obtained with the latter were very misleading, for in the first place unlike Wright's it uses a roof prism the left side of which reflects the image of the right pharynx and vocal cord, and the other that of the opposite, thereby picking up the two images and merging them in such a way as to wipe out the delicate shadows which show the true relationship of the false and real cords. Some singing teachers who have been using the latter have thereby been led to draw many false conclusions. Then too the spherical distortion is such as to make it inadvisable for use in scientific observation, though serving the medical doctor and average teacher very well for gross examination purposes. Observations made therewith should also be carefully checked with some such instrument as the *fonofaryngoskop*, for since the other apparatus in question uses a lens which very much reduces the image, and since that reduction shifts very radically as the larynx falls, one is likely to be seriously misled. For the larynx is constantly rising and falling during normal speech. But its principal objection lies in the fact that the lateral diameter is entirely too large being as great as that of my *fonofaryngoskop* which you note is as wide as my thumb; and with a piece of apparatus that wide in the mouth one can only hum, and correctly articulate but two or three vowels such as "Eh" (ɛ) and none of the lingual consonants which make up such a large part of speech. The first four mentioned of mine avoid that objection.

This phase of our investigation has been very protracted, having been begun before the war. The results are reported in detail in "Speech and Voice" Macmillan. Suffice it to say they show the whole interior muscular surfaces of the larynx are in constant and wonderous play during the changes not only of voice but of vowel quality. The positions taken are constantly varying. And surely these characteristic changes cannot be without significance. The *cushion of the epiglottis* (*pulvinar*) or *tubercle* and the *cartilages of Wrisberg* which have heretofore been totally disregarded, seem to play the most important part. To mellow the tone,

and even to lower the pitch excessively, they regularly approximate thus forming a deadening cap over the vocal cords. And one notes a characteristic progressive approximation of this type taking place as one passes through the series of vowels *i* (peep); *I* (pip); *e* (pape); ɛ (pep); *æ* (pap). You may see this in my photographs[13] of the vocal cords in "The Vowel." Then the tip of the epiglottis itself goes back toward the pharynx to shut out the view of the cords completely in *a* (pop—father).

The false cords or ventricular bands, which are often spoken of as not being subject to contraction, are shown to be playing a very vital part in creating speech and voice quality distinctions. They regularly contract and close in coughing, and in swallowing, as does the tubercle in its approximation towards the cartilages of Wrisberg. The false cords also approximate in the German "Glottal Stop". Also in the creation of excessively guttural quality in the voice; then even the mucus in between is made to vibrate. All the muscular surfaces on the interior of the larynx contract during the production of excessively high pitched strident tones, and their (impingement through the intervention of the ventricula bands), on the vocal cords themselves forces the two edges of these glottal wedges to clash together in cymbal fashion.

The cords themselves are seen to become very sharp edged in the production of what the singer calls "bright" quality, "white" tones, the vowel *i* ("*ee*") and under other circumstances. On the other hand they become round edged and blunt as the tonal quality becomes mellower and one passes to such vowels as *I* (pip) and ɛ (pep). And the slit between these rounded edge glottal wedges (vocal cords) spreads as the tone passes into what is called a "soft falsetto." An excessive spread produces a "hollow" barrel-like quality in the voice.

We might continue this paper indefinitely. But I have said enough to indicate that a physiological study has much to offer in a consideration of the question as to *what causes vowel and voice quality differences*. I believe also, that I have cited enough evidence to prove that the cavity tone theories merely skim the surface. And to prove that we are not justified in disregarding the function of the hard and soft surfaces which line the cavities. Or in minimizing the function of the vocal cords and interior larynx including the epiglottis which of themselves may do much to change both qualities. The latter phase of my investigation is continuing under a grant from the Carnegie Corporation, and the auspicies of The American Academy of Teachers of Singing, to both of which organizations I am much indebted, for making the present research possible.

THE FREQUENCY OF OCCURRENCE OF SPEECH SOUNDS IN SPOKEN ENGLISH

By Norman R. French and Walter Koenig, Jr.
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This paper gives some of the results of a study of the kind and frequency of occurrence of the words of telephone conversations, and the analysis of these words into the simple sounds of speech. The material for this study was obtained on typical toll circuits terminating in the City of New York. The process of noting the words of the conversations was carried out in the following manner. During one week the observer recorded nothing but the nouns used, during another week she recorded only the verbs; similarly for the other parts of speech. The observations were continued until data had been obtained on 500 conversations for each part of speech. With this method the results are, of course, not identical with those which would have been obtained had the entire conversations been recorded. It is likely, too, that some words were missed by the observer.

Chart 1

Words of 500 Telephone Conversations

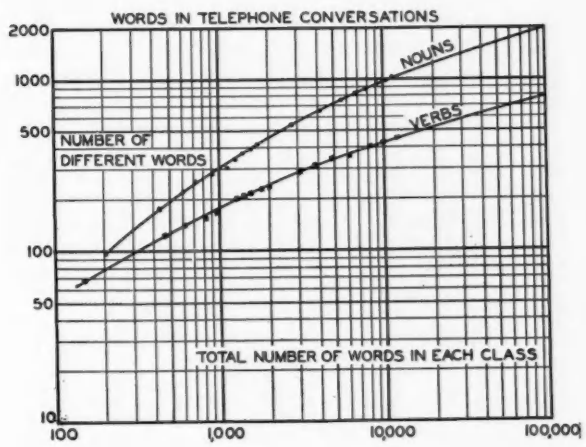
Parts of Speech	Number of Words	
	Total	Different
Nouns	11,660	1,029
Adjectives and Adverbs	9,880	634
Verbs	12,550	456
Auxiliary Verbs	9,450	37
Pronouns*	17,900	45
Prepositions*	12,400	36
Articles*	5,550	3
Total	79,390	2,240

* Derived from data on less than 500 conversations.

Chart No. 1 shows the total number of words and the total number of different words recorded for each part of speech separately. One of the striking features of this table is that of the 80,000 words recorded, only 2240 were different words. The various forms of the words, such as the plural form of the nouns, the different tenses of the

verbs, and the comparative and superlative forms of the adjectives were not counted as separate words in preparing this chart although they were recorded. One exception to this is that each form of the auxiliary verbs "be," "can," "may," etc., was counted as a separate word. If all modifications are counted as different words the number of different words is increased to 2822; but even on this basis less than 4 per cent of the total words were different words. It is probable that this percentage would be still smaller were it not for the fact that the table contains no data for the "hello" and "goodbye" stages of the conversations, during which a few stock words are used. Also certain classes of words like "yes" and "no" were not recorded as in general they seem to be merely symbols signifying that a person is still listening or understands the remarks made.

CHART 2



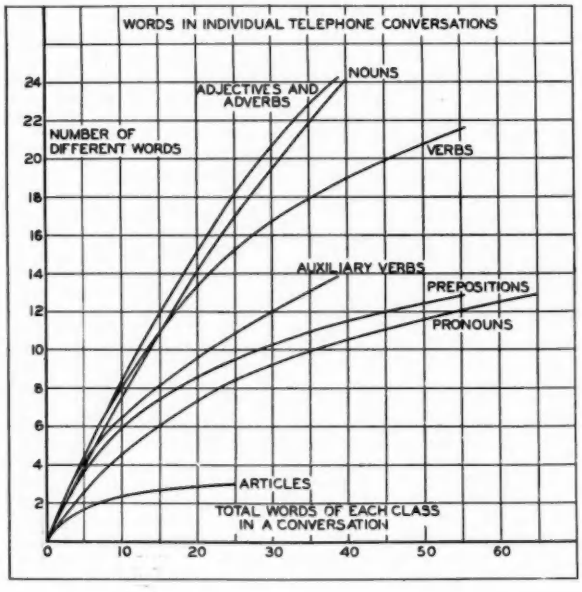
The individual lines of this chart merit more discussion than can be given here. It will be noted, however, that while the nouns constitute approximately only 15 per cent of the total words, yet they contain nearly half of the different words. It seems surprising that of the nearly 12,000 nouns recorded, only 1029 were different ones.

Chart No. 2 shows how the number of different words increases as additional words are recorded. For example, among the first 1000 nouns recorded, there were 300 different nouns, while observations on ten times as many nouns increased the total number of different nouns to 1000. Projecting the curve indicates that even with a record of 100,000 nouns there would be of the order of only 2000 different nouns.

For the verbs, the first 1000 recorded contained less than 200 different verbs. The indications are that a record of 100,000 verbs would show not over 800 different ones. These figures demonstrate the intensive use made of a few words of our language.

With regard to individual conversations the intensive use of the words used is not quite so striking. Chart No. 3 shows for the different parts

Chart 3



of speech separately the average number of different words in conversations having different total numbers of words. For example, in conversations in which 40 nouns were used, the average number of different nouns was 24. In conversations with 40 verbs the average number of different verbs was only 19.

Making use of the curves of this chart and the relative occurrences of the different parts of speech determined from the data of the first chart, the next chart (No. 4) has been prepared to show the number of different words of all classes in conversations of different durations. Thus, for a conversation of 100 total words, there are approximately 60 different ones, while for a conversation having 600 words, the number of different words is increased to about 160. Six hundred words represent a conversation having a duration of about four and one-half minutes.

CHART 4

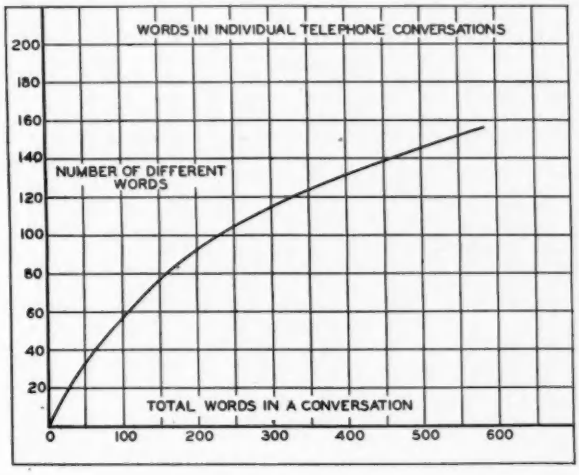


CHART 5

TWENTY FIVE COMMONEST WORDS OF ENGLISH

WRITTEN MATTER DEWEY'S ANALYSIS	TELEPHONE CONVERSATION WORDS	NUMBER OF CONVERSATIONS	TOTAL OCCURRENCE
THE	I	467	3999
OF	YOU	499	3540
AND	THE	496	3110
TO	A	487	2060
A	ON	458	2046
IN	TO	472	1942
THAT	THAT	397	1792
IT	IT	417	1615
IS	IS	419	1506
I	AND	391	1363
FOR	* GET	393	1360
* BE	* WILL	402	1305
* WAS	OF	396	1190
* AS	IN	408	1170
YOU	HE	297	1115
* WITH	WE	294	1100
HE	* THEY	253	913
ON	* SEE	328	887
HAVE	HAVE	367	883
BY	FOR	330	823
* NOT	* KNOW	338	753
* AT	* DON'T	301	640
* THIS	* DO	302	638
ARE	ARE	293	618
WE	* WANT	297	599
		TOTAL	36948

* WORDS NOT COMMON TO BOTH LISTS

The next chart (No. 5) perhaps is of the greatest interest in that is shows the particular words which are characteristic of telephone conversations. The second column of this chart shows the 25 commonest words recorded. The third and fourth columns show the number of conversations in which each word was used and the total occurrence of each word. Each of these 25 words occurred in practically 60 per cent of all the conversations and together they constitute over 45 per cent of all the words recorded.

As might be expected, the words "I" and "you" occur more frequently in telephone conversations than any other words, and constitute over 9 per cent of the total words recorded. It is interesting to note that while "you" occurred in all but one of the 500 conversations there were 33 conversations in which the people were sufficiently modest, perhaps, not to employ the word "I." However, the total occurrence of "I" exceeds that of "you" by about 450.

In the first column of this chart are given the commonest words in written matter according to Dewey's analysis*; the starred words of these lists are words which do not occur in both lists. The differences between these two lists indicate in a quite definite manner the differences which exist between written matter and conversation. For example, "I" and "you" are tenth and fifteenth in order of occurrence in written matter while they are first and second in conversation. The other chief difference between the two lists is the greater preponderance of verbs in the conversational list; it contains "get," "will," "see," "know," "don't," "do" and "want," which are not among the 25 commonest words of written matter.

On the basis of the 50 commonest words, which could not be conveniently shown in one chart, the differences between the two lists are still more marked. Only 29 words are common to the two lists. Of the 21 words of the conversational list which do not occur in the other list, 13 are verbs; in addition to those shown in this chart are "go," "tell," "think," "say," "can" and "call." Among the 50 commonest words of telphone conversation are the nouns "day," "thing," and "morning." No nouns occur among the 50 commonest words of written matter.

Before leaving this chart let us emphasize again the extent to which these words are utilized in conversation; the 25 words on this chart plus "go," "tell" and "with" made up one-half of the total words recorded. For the more diversified English analyzed by Dewey, 69 words were required to account for one-half of the words used.

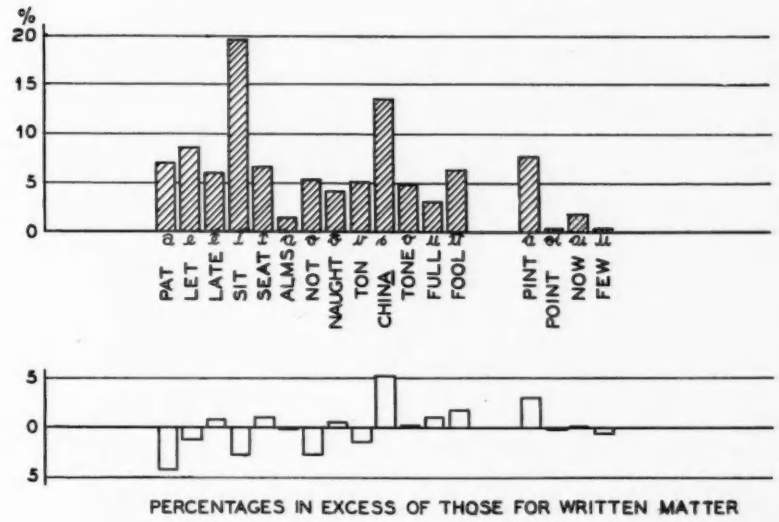
* Godfrey Dewey, "Relativ Frequency of English Speech Sounds."

Before passing to the analysis of the words into the simple sounds of English it may be well to note that the differences which have just been pointed out between the two lists of the commonest words are bound to result in materially different occurrences for certain of the sounds in conversation and in written matter.

The following charts show the relative occurrence of the simple vowel and consonant sounds as determined by our analysis of the 80,000 words recorded. The phonetic alphabet used in the tables is the one recommended by Dewey in his book. It embraces 24 consonants, 13 vowels and 4 diphthongs. But instead of adopting a dic-

CHART 6

RELATIVE OCCURRENCE OF THE SIMPLE VOWEL SOUNDS IN TELEPHONE CONVERSATION



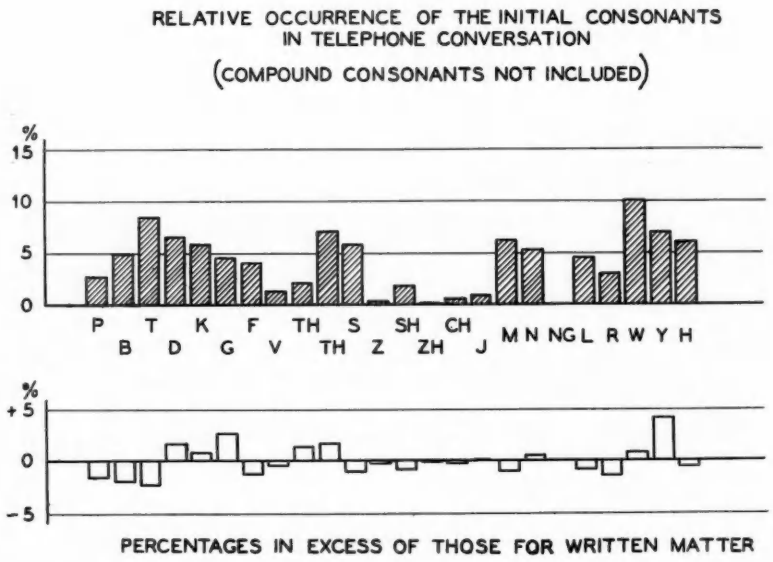
PERCENTAGES IN EXCESS OF THOSE FOR WRITTEN MATTER

tionary as an authority for the pronunciation, our analysis was carried out by assigning to each sound in a word the pronunciation which we regarded as being a typical pronunciation for reasonably enunciated speech, because we were interested in the actual rather than the theoretical occurrence of the different sounds. An example of this is the word "information." A dictionary pronunciation is "in-for-mation," while "in-fer-mation" seemed to us the more usual pronunciation. Another illustration is the word "difference," which to us seemed to lose the second syllable in conversation—"diffrence." Our classification of the sounds, therefore, is not on the same basis as Dewey's. With this

difference in mind, it is of interest to compare the results of our analysis with the results obtained by Dewey for written matter.

Chart No. 6 shows the frequency of occurrence of the simple vowel sounds in telephone conversation and also the amounts by which our figures of percentage occurrence differ from the percentages for written matter. The differences below the line indicate that our analysis showed a smaller occurrence than Dewey's, and those above the line, a greater occurrence. In preparing the following tables, the words "the" and "a" were omitted because their pronunciation is so indefinite and varied.

Chart 7

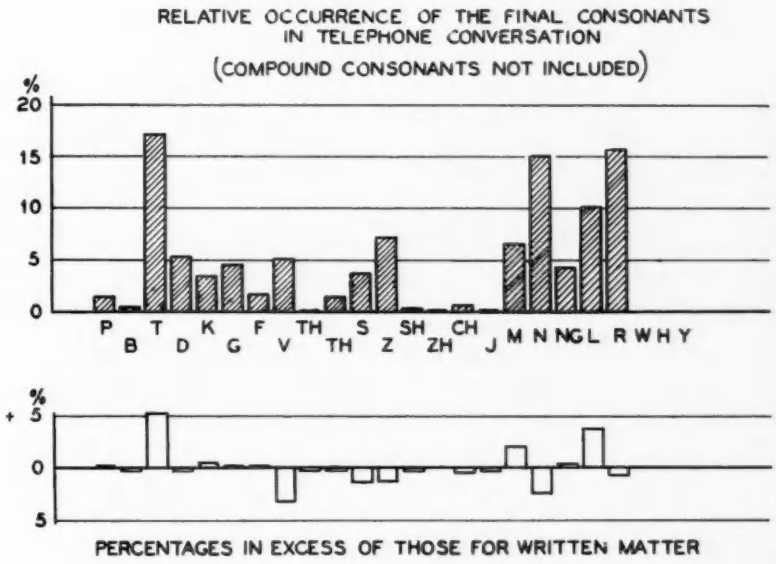


The lower part of the chart shows a close agreement between the results of our analysis and Dewey's for about half of the sounds; for the other sounds it shows large differences. A qualitative inspection of the commonest words of conversation and of written matter indicates that these differences are due primarily to the differences between the words of telephone conversation and written matter. For instance the excess in the sound "i" as in "pint" is due largely to the greater frequency of the pronoun "I" in conversation. The excess in the "u" as in "fool" is due to the pronoun "you." The excess in the "e" as in "seat," is due to the large occurrence compared to written matter, of several words, among them the verb "see" which is the 17th word in our list, but the 94th word in Dewey's. "See" is a good illustration of

a word other than a pronoun which is very common in speech but not in written matter. The unaccented vowel as in "China," however owes its high occurrence largely to the great frequency of certain prepositions and auxiliary verbs which are almost always unaccented in speech and, therefore, seldom receive their true pronunciation. The large occurrence assigned to this unaccented vowel is at the expense of several other vowels which appear low in the analysis compared to Dewey's namely the vowels of "pat," "not," and "ton."

Turning now to Chart No. 7 which shows the relative occurrence of the initial consonants, it will be noticed that the sounds most fre-

CHART 8



quently used were the "W" and the "T" sounds, which is also true in Dewey's analysis. The largest difference will be noticed in the "Y" and "G" sounds. Here again, the differences can be explained on the basis of the 50 commonest words. The verbs "get," "go," and "give," and the adjective "good," all occur higher in our list than Dewey's first "G" sound. As for the "Y" sound, the pronoun "you" is sufficient to account for the difference.

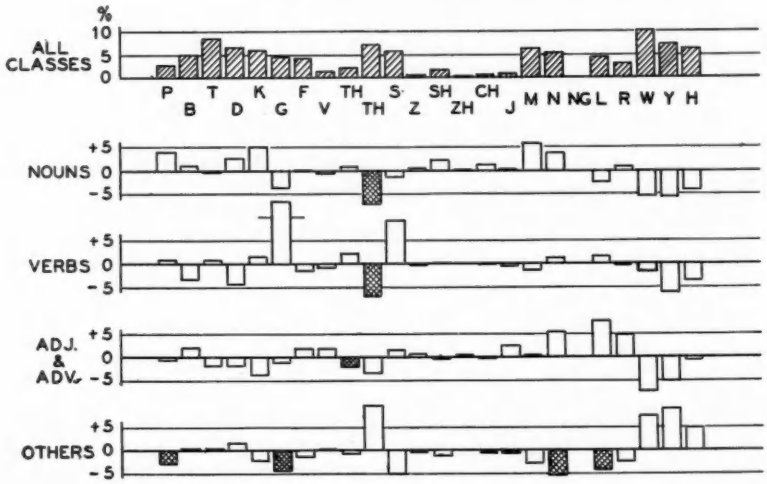
Chart No. 8 shows the relative occurrence of the final consonants in conversation. Here the "T" and the "R" sounds are the most frequently used. For the "R" sound our analysis agrees with Dewey's, but for the "T" sound our result is much higher than his. Greater

occurrence in the first 50 words of "it," "that," "what," and in addition "get" and "about" is largely responsible for this. In the next 50 words occur such characteristically conversational words as "right," "let," "wait," and "night." Similarly with the "L" sound, we find a large occurrence of the words "will," "tell," "call."

For our purposes it was of interest to investigate whether the distribution of speech sounds determined for all words is characteristic also of the words of the different parts of speech—nouns, verbs, etc. Chart No. 9 gives the results of such an analysis, and shows that the

CHART 9

RELATIVE OCCURRENCE OF THE INITIAL CONSONANTS IN WORDS OF DIFFERENT PARTS OF SPEECH



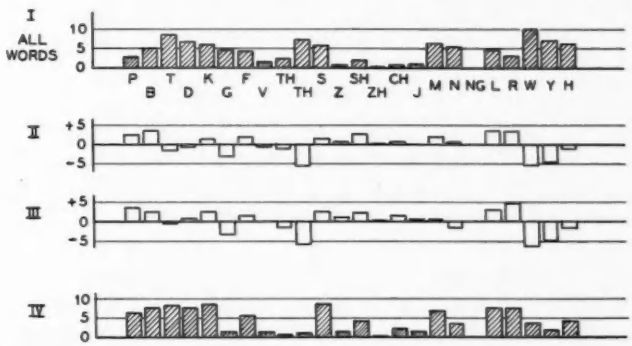
distribution of the initial consonants varies greatly for different classes of words. We have chosen the initial consonants as an illustration, the top part of the chart being a repetition of the one shown before. Now when the nouns are analyzed alone, the results are very different, as shown by the second line, where the differences are plotted. The third line is for verbs alone, the fourth for adjectives and adverbs. The last is for the "other" classes which include pronouns, prepositions, conjunctions, and auxiliary verbs such as "am," "was," "is," "has," "will," etc. These auxiliary verbs were not included among the verbs, although some of them are often used as verbs. The method of taking data did not make possible the distinction between "have" as a verb and "have" as an auxiliary verb. It will be noticed that the differences

between these classes are great. For instance, the shaded areas below the lines indicate cases where a particular sound was almost completely lacking in one class of words. Notice that the "TH" sound as in "then" is almost lacking in the nouns and verbs, and is low in the adjectives, but depends for its total occurrences on the minor classes of words. The same is true of the "Y" and "H" sounds. The "G" sound, on the other hand, is almost lacking in all classes except the verbs. Here the influence of such words as "get," "got," "give," and "go" is evident.

In considering the occurrence of speech sounds in telephone conversations from the point of view of their contribution ot the ease or difficulty of carrying on conversations over the telephone, we were

CHART 10

RELATIVE OCCURRENCE OF THE INITIAL CONSONANTS IN TELEPHONE CONVERSATIONS



I DIFFERENCES RESULTING FROM OMISSION OF WORDS USED IN MORE THAN 20% OF THE CONVERSATIONS
II " " " " " " " " " " 10% " " "
IV OCCURRENCE FOR WORDS USED IN LESS THAN 10% OF THE CONVERSATIONS

interested to see how the occurrence of the speech sounds for the less frequently used words compared with the occurrence for all words.

Chart No. 10 shows the results of such an analysis for initial consonants. The top line once more is a repetition of the previous ones. The next line shows the differences that result when all words which occurred in more than 100 out of the 500 conversations observed upon are ommitted from the analysis. The differences, you will notice, are great; in many cases they are comparable with the original magnitudes, although the number of different words thus omitted was only about 100. Next we removed about 80 additional words, namely, those which occurred in more than 50 out of 500 conversations. The third line shows the difference between that result and the original;

the differences are just about the same as in the previous line, indicating that the first hundred words are the ones that most decidedly influence the total occurrence of the separate speech sounds. The last line shows the occurrence of each speech sound after the 180 commonest words are removed from the analysis. That is, line I plus line III equals line IV. Little is left of the occurrence of the sounds "G," "TH," "W" and "Y," while the occurrence of the sounds "P," "SH," "CH," and "R" is about doubled.

EFFECTS OF DISTORTION UPON THE RECOGNITION OF SPEECH SOUNDS

By John C. Steinberg
Bell Telephone Laboratories

Introduction

The waves of speech sounds are characterized by three quantities, amplitude, frequency and phase, all three of which are essential to the correct recognition of the sounds by an auditor. In general, the term distortion refers to relative changes in one or more of these quantities, in the process of transmitting the sound waves from speaker to auditor. Recognition in the sense used here refers to the correctness with which an auditor identifies the sound that he hears as being one, or some combination, of the fundamental speech sounds, when the combinations have no thought or meaning.

Distortion not only affects the recognizability of speech sounds but the tonal character or naturalness of the sounds as well. The former aspect admits of a quantitative determination by means of the so called articulation test.[1] The latter aspect, is considerably less definite and does not appear to be very simply related at least, to the recognizability. This paper is chiefly concerned with the effects of some of the more common types of distortion, and also the effects of the presence of extraneous sound waves, such as noise, upon the recognition of speech sounds.

In an articulation test a speaker utters the various speech sounds and an observer writes down the sounds which he hears. The observed sounds are compared with the called sounds and the percentage of the called sounds that were correctly recognized obtained. This percentage which is called the articulation is taken as a measure of the recognizability of the speech sounds.

The sounds that are used are shown in Table I. In order to call the sounds they are combined into syllables of the consonant-vowel-consonant type. To do this the initial consonants, the vowels, and final consonants are written on cards and are placed in three separate boxes.

[1] Articulation Testing Methods, H. Fletcher and J. C. Steinberg, Bell System Technical Journal, Oct. 1929.
Speech and Hearing, H. Fletcher, D. Van Nostrand Co., 1929.
Nature of Speech and its Interpretation, H. Fletcher, Jour. Frank. Inst., 1922.

A syllable is formed by drawing a card from each box. All of the combinations except those stated in Table I, and those which represent slang in English are used.

TABLE I

SPEECH SOUNDS FOR NEW STANDARD TESTING LISTS*

Initial Consonant	I. P. A.	Key Word	Vowel	I. P. A.	Key Word	Final Consonant	I. P. A.
b			a	[ɑː]	father	b	
d			a			d	
f			ā	[eː]	fame	f	
g		go	ā			g	
k			a′	[æ]	fat	k	
l			a′			l	
m			e	[ɛ]	get	m	
n			e			n	
r			ē	[iː]	greet	r	
p			ē			p	
s			i	[I]	tin	s	
sh	[ʃ]	ship	i			sh	
th′	[ð]	this	o	[ʌ]	but	th′	
th	[θ]	thin	o			th	
t			ō	[oː]	go	t	
v			ō			v	
ch	[tʃ]	church	u	[U]	full	ch	
z			u			z	
j	[dʒ]	judge	ū	[ūː]	rule	j	
h			ū			zh	[ʒ]
w			o′	[ɔː]	haul	ng	[ŋ]
y	[j]	yawl	o′			st	

Note: Final r and ng are used in the list only when they occur in combination with the following vowels:

a′r	(as in carry, paragraph	a′ng	(as in bang, sang)
ar	(as in are, far, tar)	eng	(as in geng, e as in ten)
er	(as in bury, ferry, verify)	ing	(as in sing, wring)
ir	(as in spirit)	ong	(as in sung, hung)
or	(as in her, utter, fir)	ung	(as in gung, u as in took)
o′r	(as in for, lore)	o′ng	(as in long, wrong)
ūr	(as in your, sure)		

* The symbols for the sounds are those of the International Phonetic Association's Alphabet. See Pronunciation of Standard English in America, Krapp, Oxford University Press, 1919. See also Revised Scientific Alphabet, Funk & Wagnalls Dictionary.

The syllables are spoken as parts of the introductory sentences shown in Table II, where the syllable to be called is written in the

column that is marked "called." The observers write down the observed syllables in the "observed" columns of blank sheets. The introductory sentences are used in order to simulate connected speech

TABLE II

ARTICULATION TEST RECORD

DATE 3-16-28 SYLLABLE ARTICULATION 51.5%

TITLE OF TEST PRACTICE TESTS CONDITION TESTED 1500~ LOW PASS FILTER

TEST NO. 10 OBSERVER W.H.S.

LIST NOS. 5-9-37 CALLER E.B.

NO.		OBSERVED	CALLED	OBSERVED	CALLED	OBSERVED	CALLED
1	THE FIRST GROUP IS	mav	náv	póz	po'th	Kōb ✓	Kōb
2	CAN YOU HEAR	pōch ✓	pōch	nēz	nēzh	shēth	siz
3	I WILL NOW SAY	seng ✓	seng	jóch ✓	jóch	fūch ✓	fūch
4	AS THE FOURTH WRITE	chūd ✓	chūd	thám ✓	thám	thōl ✓	thōl
5	WRITE DOWN	run ✓	run	hab ✓	hab	poth ✓	poth
6	DID YOU UNDERSTAND	chiz	kiz	def	doth	wám ✓	wám
7	I CONTINUE WITH	foz	fozh	chech	chej	gūm	gūn
8	THESE SOUNDS ARE	lól ✓	lól	lun	lon	nāsh	nāth
9	TRY THE COMBINATION	jās	zhāth	shāl ✓	shāl	vóg ✓	vóg
10	PLEASE RECORD	tháth	thásh	muz ✓	muz	lung	long
11	WRITE THE FOLLOWING	wūr ✓	wūr	led	bēd	diz	dizh
12	NOW TRY	yāp ✓	yāp	wif ✓	wif	Kak	tak
13	THIRTEEN WILL BE	mad	maj	gōst ✓	gōst	thár	zhár
14	YOU SHOULD OBSERVE	bēch	bēk	thav	sāv	must ✓	must
15	WRITE CLEARLY	gēm	dēm	Kōf ✓	Kōf	yód ✓	yód
16	NUMBER 16 IS	theb	veb	rág ✓	rág	jet ✓	jet
17	YOU MAY PERCEIVE	jok	jost	thip ✓	thip	rēp	rēj
18	I AM ABOUT TO SAY	gaf ✓	gaf	yar ✓	yar	thēp	hēp
19	TRY TO HEAR	hus ✓	hus	zhūt ✓	zhūt	—	chuv
20	PLEASE WRITE	hiv	thith	Kūk	tūk	thef	thesh
21	LISTEN CAREFULLY TO	tōg ✓	tōg	fung ✓	fung	bās ✓	bās
22	THE LAST GROUP IS	shót ✓	shót	thev	vesh	thóf	shaf

as nearly as possible. In making a test, a list of 66 syllables is called by each of eight speakers, (4 men and 4 women), and four observers per caller are used to record the syllables.

The percentage of the number of called syllables that are correctly observed is called the syllable articulation. The articulation of a specific fundamental sound, such as "b" for example, is the ratio, expressed as a percentage, of the number of times it was correctly recognized to the number of times it was called. In discussing the articulation values of the individual sounds it is convenient to collect the thirty six sounds into five groups, namely, long and short vowels, nasal, stop and fricative consonants, and for each group to deal with an average articulation of all of the sounds in the group. The sounds composing each of the five groups are shown below.

Long Vowels—a ā ē ō o′ ū
Short Vowels—a′ e i o u w y
Nasal Consonants—l m n ng r
Stop Consonants—b ch d g h j k p t
Fricative Consonants—f s sh st th th′ v z zh

In order to gauge the effects of distortion, it is necessary to have as a reference, a system which is as nearly free from distortion as possible. Under suitable conditions, air transmission from speaker to observer is a system which is reasonably free from distortion. It is difficult, however, to change and control such a system in desired ways. For this reason, an electrical system has been built in the Laboratories which is a very good simulation of air transmission for speech sounds. The Master Reference System for Telephone Transmission[2] is similar in many respects to this system. By the use of equallizing networks

Fig. 1. *Transmission System.*

the system was arranged so as to transmit the frequency range from 80–15000 cycles without appreciable distortion. By means of attenuators and filters, the received speech level and the transmitted frequency range could be changed in known ways. Other apparatus producing specific types of distortion could also be introduced into the system. The arrangement of this system is shown in Fig. I. It contains essentially a condenser transmitter and its associated amplifier with an equallizing network, terminals for inserting various kinds of dis-

[2] Master Reference System for Telephone Transmission, W. H. Martin and C. H. G. Gray, Bell Tech. Jour. July 1929.

torting networks, an attenuator for changing the received speech level, and a receiver with its associated amplifier.

Effect upon Articulation of Changing the Level of the Received Speech

The recognition or articulation depends upon the level of the speech that reaches the observer's ear. It is difficult to understand not only very faint speech but very intense speech as well. Fig. 2 shows the effect upon syllable articulation of changing the sensation level of speech through a wide range of values. In making these tests the system was arranged so that no distortion was introduced, and the level was changed by means of the attenuator. An average attenuator setting

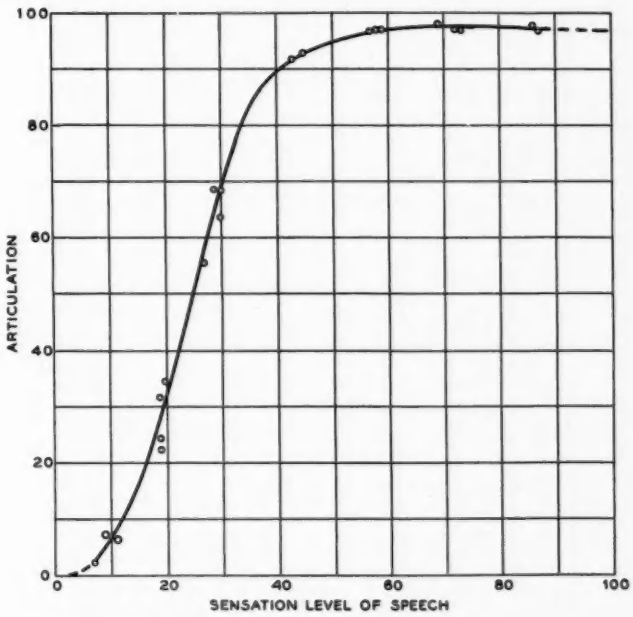


Fig. 2. *Effect of speech level on syllable articulation.*

for the threshold of audibility was determined for the various speakers and observers. In determining threshold the attenuation was changed in steps of 5 db. Usually a point was found where very faint speech could be heard. When 5 db more attenuation was inserted, the peaks of speech were so faint and intermittent as to escape momentary notice. In other words a point was found where an additional 5 db change made, by comparison, a very marked change in the level of the

received speech. The setting for threshold was taken as half way bebween these two points. It was found that the average value (an average of about 50 observations) obtained in this way could be reproduced to within about 2 db. The speech used in making these observations was the articulation syllables inserted in the introductory sentences.

It is evident from Fig. 2, that the articulation increases very rapidly as the sensation level is increased until a value of about 50 db is reached. The articulation remains approximately constant as the level is increased above 50 db, and decreases slightly for sensation levels greater

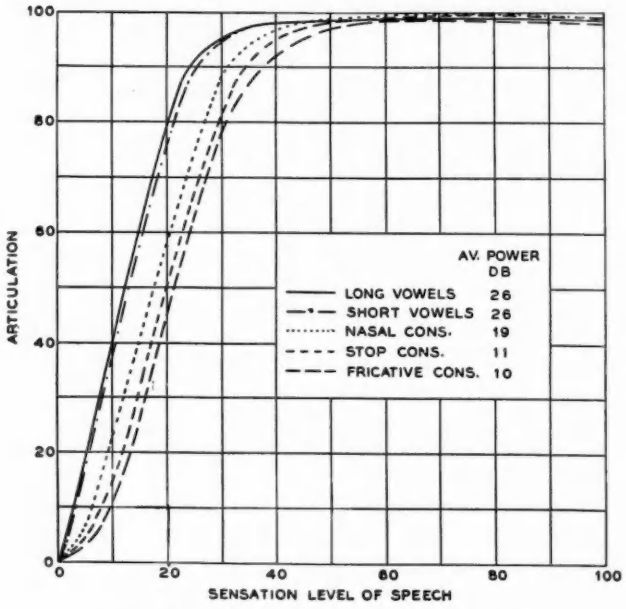


Fig. 3. *Effect of speech level on sound articulation.*

than 70 db. The data shown were taken from four series of tests that were made at various intervals over a period of about a year.

Oscillograms[3] of speech waves have shown that the sounds of speech are uttered with inherently different intensities, the vowels being the most intense, the fricative consonants being the least intense. Fig. 3 shows the effect of received speech level upon the articulation of the five groups of sounds discussed above. These curves were obtained by analyzing the mistakes made in the syllable tests of Fig. 2. It appears

[3] Sounds of Speech, I. B. Crandall, Bell Sys. Tech. Jour., Oct. 1925.

The Power of Fundamental Speech Sounds, C. F. Sacia & C. J. Beck, Bell Sys. Tech. Jour. July 1926.

that as the speech level is increased from threshold, the vowels are the first to be recognized and are followed in order by the nasal stop and fricative consonants. The lowest speech levels tested were about 10 db where articulations of an order of 35% were obtained for the vowels and only 10% for the fricative consonants. This indicates considerable difference in the inherent levels of these groups of sounds. It is difficult, however, to compare the levels of the various sound groups at this speech level because of the large probable error in testing at speech levels below 10 db. Also at these levels, the consonants may be interpreted from their effects upon the beginning and ending of the vowel sounds, or at least various groups such as stop and fricative consonants may be identified. When this is the case, articulation values below 20% are not significant of level differences because straight guessing would give articulations of this order since there are only 5 to 10 sounds in a group. The data indicated, however, that the threshold for average speech corresponds roughly to the vowel threshold and that the level articulation curves for the other groups converge to this point as shown in the figure.

The differences in articulation for speech levels around 25 db appear to be due to differences in the inherent levels of the sound groups. For example, the vowel articulation at 25 db is about 90%. The nasal consonant group must be increased 6 db in order to obtain this articulation value, and the stop and fricative consonants must be increased by 10 and 13 db respectively. These values correspond closely to the db differences in the relative phonetic powers[4] given in Fig. 3. In other words, when the sensation level of the speech is below 50 or 60 db, the articulation of the various sounds depends primarily upon the inherent powers with which they are uttered, the faint sounds being the more difficult to recognize, the loud sounds being the easier. At the higher levels, the nasal consonants are the easiest to recognize, the optimum articulation being 99.7%. These are followed in order by the stop consonants 99.3%; short vowels, 99.2%; long vowels, 98.8%; and the fricative consonants, 98.5%.

Effect upon Articulation of Removing Portions of the Speech Frequency Range

The oscillograms mentioned above, also show that the principal frequencies found in different speech sounds may be quite different. For example, the frequencies found in "s" would hardly be found in

[4] Speech and Hearing, H. Fletcher. (See Table X, p. 74).

ū (as in rule). In order to determine the importance of various frequency ranges for the recognition of speech sounds, electrical filters were inserted in the system shown in Fig. 1. Two sets of filters were used, one set, (low pass filters), eliminated all frequencies above a chosen cut-off frequency and uniformly transmitted the remainder of the range; the other set, (high pass filters), eliminated all frequencies below a chosen cut-off frequency. The eliminated region was attenuated 90 db or more as compared with the transmitted region, so that, for the speech levels that were tested the attenuated region was always below threshold.

Fig. 4 shows the effects upon articulation of changing the cut-off frequency of these filters over the speech frequency range. The curves marked L.P. mean that only frequencies below the cut-off frequency

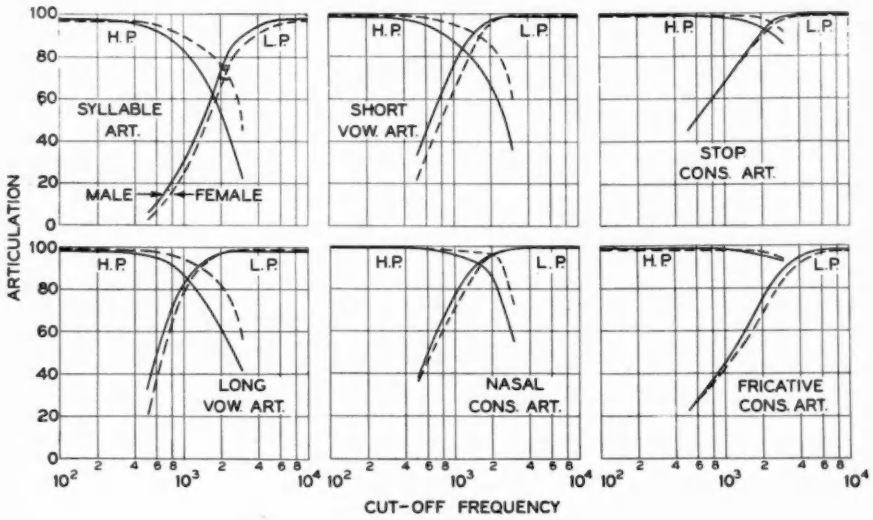


Fig. 4. *Effect of low and high pass filters.*

were transmitted; for the H.P. curves only frequencies above the cut-off frequency were transmitted. The frequency range through which the curves bend down is the range of importance to the recognition of the sounds. For the case of syllable articulation, which involves all of speech sounds, this range extends from about 200~ to 8000~. Each of the various groups of sounds does not require this whole range, the regions of importance being as follows: long vowels, 200–2500~; short vowels, 300–3000~; nasal consonants, 500–4000~; stop consonants, 750–5000~; fricative consonants, 1000–8000~.

Woman's speech appears to differ considerably from man's. Taking the speech sounds in the aggregate, syllable articulation, woman's speech is inherently somewhat more difficult to understand than man's as indicated by the flat portions of the curves. The long and short vowel sounds, however, are easier to recognize when spoken by women then when spoken by men. The nasal consonants appear to be equally difficult for both voices and the stop and fricative consonants are easier to recognize when spoken by men. The curves indicate that the frequency ranges of importance are shifted to somewhat higher frequencies in the case of woman's speech.

It will be seen that, in the case of the stop and fricative consonants, for example, the high pass filter tests indicate that these sounds have no frequencies of very great importance below 1000 cycles. However, when all frequencies above 1000 cycles are eliminated, (low pass filter tests), articulations of an order of 50 to 70% are obtained, which appears inconsistent with the high pass filter tests. At least part of the explanation of this phenomenon is that a given consonant is identified as belonging to a certain group from the nature of the beginnings or endings of the vowel sounds, and also from the transmitted portion of the sound itself. For example, the voiced consonants are rarely mistaken for the unvoiced sounds. The stop consonants are rarely mistaken for the fricative sounds. Under these conditions, the 18 consonant sounds are identified in groups of three or four sounds, so that articulations of an order of 20 to 35% may be obtained by merely writing down any sound in the group.

General Effects of Distortion

The data just discussed show that in normal speech the sounds are distributed in intensity and frequency. The primary effects of distortion may be readily understood if this fact be borne in mind. For example, if the distortion tends to increase the intensity spread between the sounds, the articulation of the less intense sounds is decreased. Similarly, any distortion which suppresses a portion of the frequency range decreases the articulation of the sounds that are characteristic of that range, or any distortion which effectively lowers the sensation level of the speech decreases the articulation of the less intense speech sounds. Various types of distortion will be considered from this viewpoint in the following paragraphs. It will be seen that although the primary effects of most types of distortion can be accounted for in this way, other factors of lesser importance also enter in.

Effect of an Extraneous Noise

The primary effect of noise upon the interpretation of speech sounds is closely associated with "masking"[5] or more conveniently "deafening." The deafening at any frequency is measured by the number of db that the sound pressure of a tone must be increased above threshold in the quiet, in order that the tone be audible in the presence of a noise. If the noise has a spectrum such that the deafening at all frequencies in the voice range is approximately constant, it is evident that the presence of the noise effectively lowers the sensation level of the speech. Fig. 5 gives the results of articulation tests that were made in the

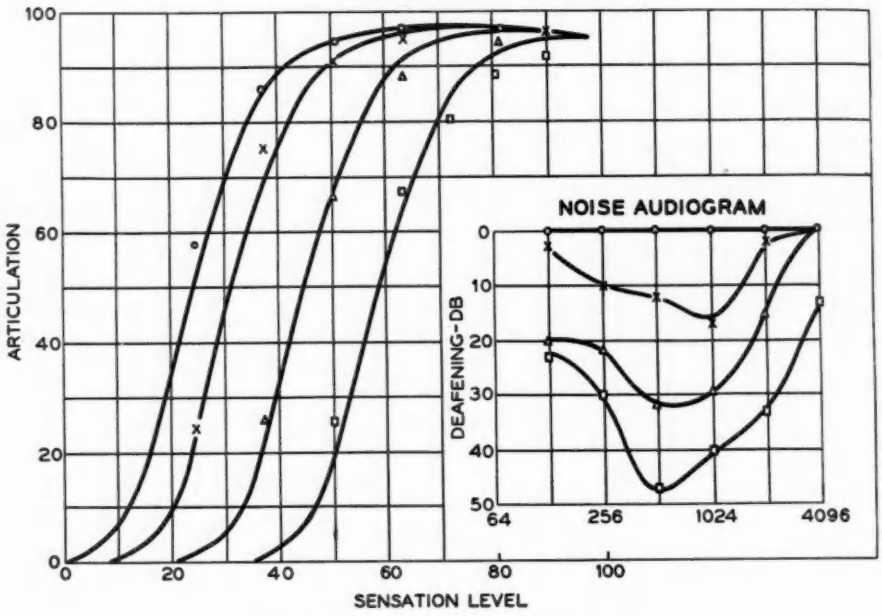


Fig. 5. *Effect of noise on articulation.*

presence of a noise that caused appreciable amounts of deafening throughout the important speech range of frequencies. The noise audiogram shows the deafening that was produced by the noise corresponding to three different intensities of the noise. Although the amplitudes of the various component frequencies were all increased in the same ratio in going from a lower to a higher intensity, it will be seen that the amounts of deafening at various frequencies did not increase uniformly.

[5] Auditory Masking, R. L. Wegel and C. E. Lane. Phys. Rev. Feb. 1924. See also Speech and Hearing, H. Fletcher, D. Van Nostrand Co. 1929.

Articulation observations vs the sensafion level of the speech are shown for the noise conditions indicated in the audiogram, namely the three noise levels and the case of no noise. The solid lines for the tests in the presence of noise were obtained by shifting the curve for the quiet condition along the x-axis to obtain the best fit of the data. It will be seen that the amounts of the shifts that were necessary are of the orders of magnitude of the deafening shown in the audiogram. For the case of small deafening, the data coincide with the curve for all of the speech levels. For noise of higher levels, however, the points all below the curve. When the speech and noise levels are both high,

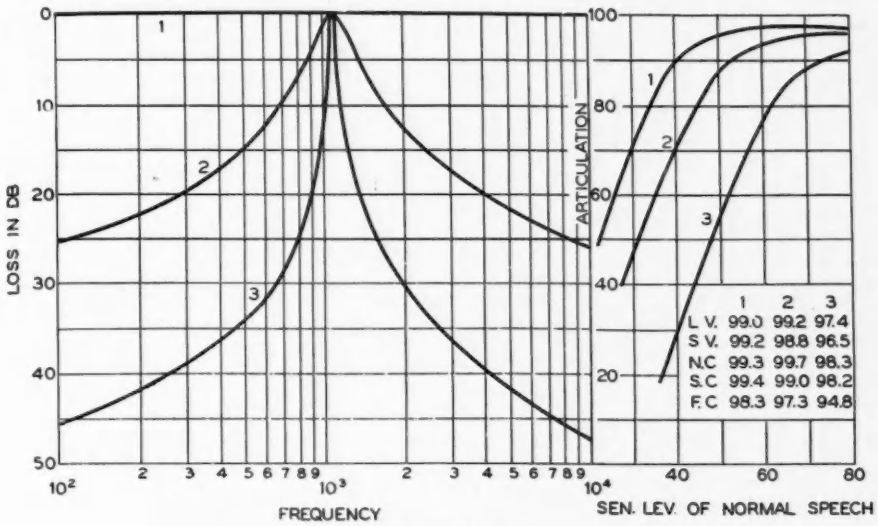


FIG. 6. *Effect of resonance type of frequency distortion.*

intermodulation takes place between the speech and the noise, which decreases the maximum value of articulation that can be obtained.

With the above exceptions, the effect of the noise amounted to a shift in the threshold of the speech. This is usually the case when the noise deafens more or less uniformly throughout the frequency range from 200 to 2000 cycles. When the principal deafening takes place at higher frequencies, for example, the deafening caused by a 2000~ tone, definite shifts of this kind are not obtained and other factors must be taken into consideration.

EFFECT OF RESONANCE TYPE OF FREQUENCY DISTORTION

In this type of distortion the transmisson efficiency is greatest at the resonant frequency and falls off on either side at a rate depending

upon the damping constants of the circuit. Fig. 6 shows the frequency characteristic of three circuits, two of which have a resonant frequency of 1100 cycles. Such a type of distortion affects the different speech sounds in different ways. Sounds having important frequencies in the region of resonance suffer distortion. Sounds having important frequencies outside the region of resonance suffer attenuation or a decrease in level. The table in Fig. 6 shows the average articulation of the groups of sounds previously discussed for the three networks, when the sensation level of the normal speech, that is, before the networks were inserted, was 80 db. It will be recalled from Fig. 4 that the important range for the fricative consonants is above 2000 cycles. Network #3 as compared with #1 introduces an attenuation of an order of 35 to 40 db in this frequency range which caused an articulation loss of an order of 4% for the fricative consonants. Reference to Fig. 3 shows that this loss is of the right order for the above attenuation. This effect would have been more striking had the level of the normal speech been lower. The important frequencies for the vowel and nasal consonant sounds lie in the region of resonance. The table shows that pronounced resonance does not seriously impair the articulation. The tonal character or quality of the reproduced speech, however, is appreciably impaired.

The curves show the syllable articulation for various sensation levels of normal speech. The decrease in the maximum articulation values appears to be caused by the distortion in the resonant region, and the shift in the straight part of the curves by the increased attenuation in the remaining frequency ranges.

Effects of Reverberation

In making tests for the effects of reverberation, the transmitter of the system was placed about three feet from the wall of a room having variable reverberation conditions. The dimensions of the room were 20 ft. $\times$ 30 ft. $\times$ 15 ft. The reverberation time[6] of the room was changed by hanging absorbing panels on the walls. Tests were made for each of the three reverberation conditions shown in Fig. 7, for the following distances between the face of the transmitter and the lips of the speaker, .12 ft., .66 ft., 3 ft. and 15 ft. To distinguish the three conditions of reverberation they are labeled by the approximate reverberation times, namely, 1.2 sec., 2.2 sec. and 4 sec. In all of the tests the speaker faced

[6] Speech and Hearing, H. Fletcher.

toward the transmitter. The level of the speech picked up by the transmitter was measured by means of a volume indicator. This instrument measures the syllabic power of the vowel sounds. The syllabic power is carried largely by the frequency range from 100 to 2000 cycles. The gain in the amplifiers between the transmitter and the receiver was set, so that, the sensation level of the received speech for the close talking condition, was 74 db. The gain in the system was then kept constant for all of the tests.

The results of the tests are shown in Fig. 8. It will be seen that the speech level falls off inversely as the square of the distance for distances

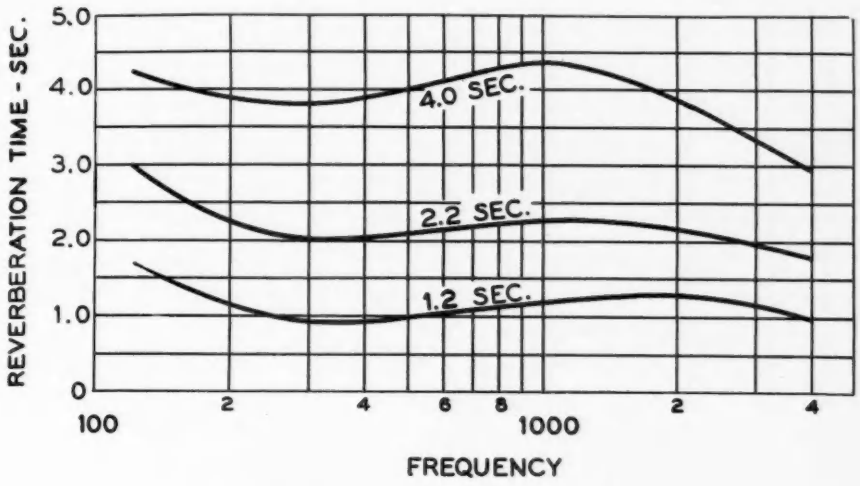


FIG. 7. *Reverberation characteristics.*

up to one foot from the transmitter. At a distances of 15 ft., however, the level is some 12 db greater than the level obtained by assuming the inverse square relation.

The articulation results for the three conditions of reverberation are shown by the dashed curves. The solid curves are graphs of Figs. 2 and 3, where sensation level has been converted into distance from transmitter by the inverse square relation. In other words, sensation level was converted into distance by means of the solid curve showing loss in speech level, the sensation level for zero loss being taken as 74 db. Since the probable error in determining the sensation level is about one db or so, the solid curves at the 15 ft. distance are uncertain within a foot or two. Gain measurements from transmitter to receiver were

not made on the system as set up for the reverberation tests, so that, the above method is the only way the two sets of tests can be compared.

The decrease in articulation shown by the dashed curves is due to two factors; the attenuation caused by the increased distance, and the room reverberation. The solid curves show approximately the decreases caused by attenuation, when the various sound groups are uniformly attenuated. The dashed curves, for the 1.2 sec. reverberation time, show approximately the decreases caused by attenuation that arises from

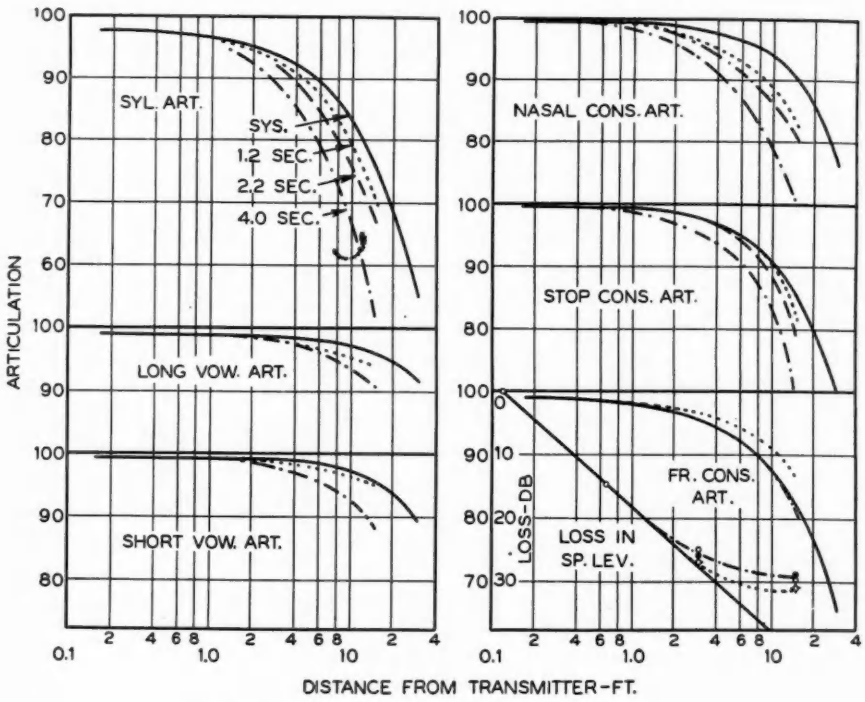


Fig. 8. *Effect of reverberation on articulation.*

increasing the distance between speaker and transmitter. A comparison of these curves with the solid curves indicates that the various sound groups were not uniformly attenuated by increasing the distance between speaker and transmitter. The groups of higher frequency sounds appear to have been attenuated a few db less than the lower frequency groups. Although this phenomenon may be related to the reflection and diffraction effects that arise when a transmitter is inserted in a sound field, the data are not extensive enough to draw definite conclusions.

It is also evident that the different sound groups were affected differently as the reverberation time was changed. The articulation of the nasal and stop consonant groups appear to have been impaired most and that of the fricative consonants least. The stop consonants are transient in character and very short in duration. It is likely that the principal factor in reducing the articulation of these sounds is distortion in the beginnings and endings due to the slower rate of growth and decay in reverberant rooms. The fricative consonants are among the faintest of the speech sounds. Their relatively good articulation indicates that the so-called "hang-over" effect, or the masking produced by the reflected speech sounds, is comparatively small. This is also indicated by initial and final consonant articulation values that are given in the table below:

TABLE III

INITIAL AND FINAL CONSONANT ARTICULATION

Rev. Cond.	Distance 15 feet			Distance 3 feet		
	Init.	Fin.	Diff.	Init.	Fin.	Diff.
Sys.	90.5	84	6.5	98.2	96	2.2
1.2	89	81	7	98.5	96.1	2.4
2.2	88	78	10	98	96	2.0
4.0	77	71	6	97	93	4.0

It will be seen that the differences between the initial and final values are about the same for the reverberation tests as was obtained for the system in which attentuation only was introduced.

It will be seen from the dashed curves of Fig. 8 that the articulation decreased as the distance between speaker and transmitter was increased in much the same way as it decreased with increased attenuation except at a more rapid rate. For distances up to 15 ft. at least, there appears to be no tendency for the curves to bend up in a way such that additional increases in distance cause correspondingly smaller losses in articulation, this in spite of the fact that the received speech level for the dashed curves is some 12 db greater than that for the solid curves. In other words the increased loudness of speech caused by the reverberant room did not contribute enough to the recognizibility of the speech to compensate for the loss caused by reverberation effects. For distances up to 15 feet at least, the articulation was appreciably less for a reverberant room than that which would have been obtained had the walls been perfectly absorbing.

It should be pointed out that the gain of the system was set such that the received speech sensation level for the close talking condition (.12 ft.) was 74 db. This level was some 20 db less than the level which the observer would have received if the speaker had spoken directly into his ear from a distance of .12 of a foot.

Other Types of Distortion

There are several other types of distortion that are occasionally met with in speech systems which will be discussed in a general way only.

Non-linear distortion arises when the output of a system is not proportional to the input. There are two cases of interest, (a) when the output increases at a more rapid rate than the input, (b) when the output increases at a slower rate than the input.

In both cases modulation products or extraneous frequencies are introduced for the intense sounds when they cover an appreciable part of the curved portion of the input vs output characteristic. The result is a decrease in the articulation of such sounds. The amount of the decrease depends upon the amount of the curvature. In addition, in the first case the intense sounds are amplified relative to the faint sounds which, as discussed above, has the effect of decreasing the articulation of the latter group of sounds. In the second case the faint sounds are amplified relative to the intense sounds. This has the effect of increasing the articulation and may, in cases where the received speech level is below the optimum receiving level, offset the decrease due to modulation effects. For optimum receiving levels, however, the net result is always a decrease in articulation when the system is free from noise.

Phase distortion takes place when the phase shift is not proportional to frequency. One type occurs when the phase shift increases with increase of frequency according to relations involving not only the first but higher powers of the frequency. In this case it is convenient to regard the received wave as made up of a distorted form of the original wave plus a transient which arrives a fraction of a second later, depending upon the phase characteristic. The main effect on articulation appears to be accounted for if the distortion is considered as due to the delaying of certain component frequencies in the original wave, so that they are effectively removed from the interpretable part of the speech sound. In other words, the effect is one of narrowing the transmitted frequency range. Thus a decrease in the articulation is brought about in the same way as narrowing the frequency range by means of filters. Although extensive tests have not been made with this type of distor-

tion, the data indicate that the delayed transients have a negligible effect upon the interpretation of the succeeding speech sounds. They may, however, cause considerable annoyance.

In some systems it is possible for frequency shift to take place. There are two types of interest, (a) multiplication of frequencies by a constant factor, (b) addition or subtraction of a constant number of cycles to all frequencies. In general the decrease in articulation is greater when the frequencies are shifted to lower values than when they are shifted toward higher values. One effect of both types of distortion is to reduce the width of the transmitted frequency band, due to shifting the frequencies from an efficient range of transmission to an inefficient range. In the first type of distortion the duration of the sounds is changed whereas in the second type the duration is unchanged. In the first type the harmonic relationship is maintained whereas in the second type this relationship is not preserved. In order to interpret speech sounds correctly it seems to be necessary to preserve the harmonic relationship and also to hold constant the frequency interval between regions of resonance set up by the vocal cavities. In these respects the two types of distortion affect the speech sounds differently. Many of the errors in interpreting the more sustained sounds of speech are traceable to these effects.

JULY 12, 1929.

A SPARK CHRONOGRAPH DEVELOPED FOR MEASURING INTENSITY OF PERCUSSION INSTRUMENT TONES

By C. N. Hickman

Research Laboratory, American Piano Company

Desiring to obtain a more accurate means of recording the loudness of each note at the time an artist is making an Ampico recording, it seemed that the logical method of procedure was to measure the velocity of the hammer at the time it struck the string.

In order to accurately record this velocity the use of a spark chronograph seemed desirable. If such a system were to be used it was obvious that it would be necessary to develop a special spark chronograph. The essential requirements of the system were as follows: First, the contacting member, which must be attached to the piano hammer shank, must be quite light, in order to prevent a change in the touch and tone. Second, the amount of energy in the primary spark must be quite low in order to eliminate noise and heavy currents. Third, the electrical system must be one which will be capable of rapid repetition. Fourth, it must eliminate the necessity of running the paper at high speeds which is the custom in obtaining space time records.

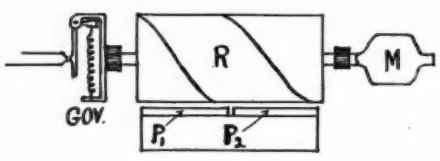


Fig. 1.

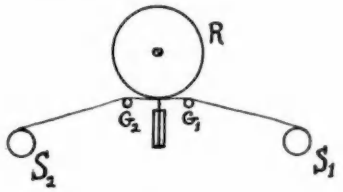


Fig. 2.

Lateral distributing spark chronograph.

The last requirement, at first, seemed the most difficult task but was eventually solved in a very satisfactory manner. A rotating drum, which was driven at constant speed by an electrical governing method, was provided with spiral knife edges for distributing the time interval sparks at right angles to the motion of the paper.

Figure 1 is a front view diagram of the spiral drum or rotor with two spark segments immediately beneath. The paper travels between the rotor and segments. (It is obvious that by making the drum sufficiently wide and adding additional segments beneath, any reasonable number of records may be obtained.)

The segment (P_1) records the first contact made by the hammer and the segment (P_2) records the second contact. The time elapsing between these two contacts is a function of the distance the rotor has turned in this interval. The spiral distributor, in this interval of time, has moved a certain distance to the right. This gives us a measure of the time elapsing between the first and second contacts. The spirals on the rotor may be at any desired angularity. In the instrument constructed an angle of 45° was used. The distance between two spirals is equal to the length of each segment.

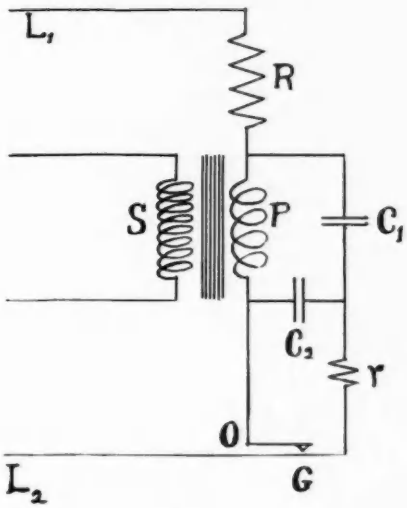


FIG. 3.

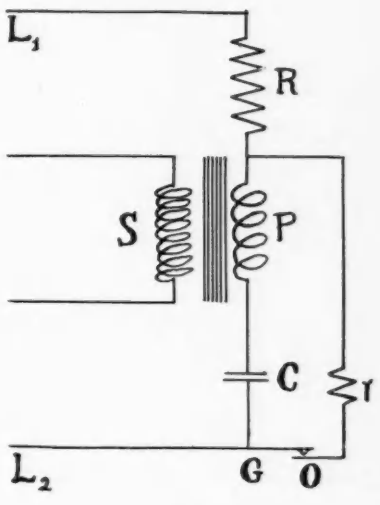


FIG. 4.

Spark Chronograph circuit.

Figure 2 shows a cross sectional view of the rotor and segments with the paper passing between them. Since the space time records are distributed at right angles to the motion of the paper, it is only necessary to run the paper fast enough to get the spark records out of the way of the succeeding records.

Figure 3 is a diagram of the electrical system for obtaining the secondary sparks used in the Aberdeen chronograph. Direct current of 220 volts is applied at the terminals (L_1) and (L_2). Condensers (C_1) and (C_2) are charged through the resistance (R) and inductance (P). The inductance (P) represents the primary of an induction coil. When the primary circuit is closed at the contacts (OG) the condenser (C_2) is discharged. The condenser (C_1) then discharges through the primary (P), inducing a secondary spark at the terminals of the coil (S).

One of the terminals of coil *S* is the rotor spirals and is common to all the secondary coils. The other terminal is one of the segments beneath the rotor.

This electrical system seemed more complicated than necessary. Another circuit was developed, which in so far as it has been tested, seems to work equally as well. Figure 4 shows this simplified circuit.

In either circuit the resistance (*R*) is made considerably smaller than that used in the Aberdeen chronograph in order to improve repetition. The same segment may be used again after an interval of less than one one-thousandth of a second.

In order to eliminate secondary sparks a damping resistance (*r*) is introduced in the primary circuit of each note. This consists of a little adjustable filament rheostat. This resistance can easily be adjusted to give a single discharge between the rotor and the segment.

It has already been stated that it was necessary to reduce the amount of primary energy to a point where both noise and heavy currents were eliminated. The heavy currents were objectionable because the contacts would soon be eaten away introducing inaccuracies. Heavy contacting numbers would have resulted in noise, unsatisfactory touch and tone.

Several methods were tested for obtaining a record of feeble sparks. One of the methods that was suggested for recording feeble sparks, was to use photographic paper, depending on the ultra-violet ray of the spark exposing the sensitized paper. This method was fairly satisfactory but objectionable for practical and commercial reasons. Later a much more satisfactory method was developed.

If the paper, which is punctured by the spark, has a coating on one side which is more or less waterproof, the spark holes may be made visible by applying a solution of dye to the coated side of the paper. Capillary attraction causes the dye to penetrate through the hole and spread slightly on the uncoated side of the paper.

Figures 5 and 6 show the manner in which the spark holes in the paper are made visible. In Figure 5, (C) represents the coating on one side of the paper and (*S*) represents the porous portion of the paper. After the spark has passed through the paper there is a hole through it, which is shown in an exaggerated manner in the middle of the figure. When dye is applied to the coated side of the paper, capillary attraction causes the dye to pass through the coating and spread in the porous part of the paper. (See Figure 6.) If the paper is left in contact with the dye for a long period of time this spot becomes very large, if, on the

other hand, the solution is wiped off very quickly, the result is a small colored spot on the porous side of the paper.

If a heavy secondary spark is used, a solution of water with any dye, which is soluble in water, is quite satisfactory. However, if a feeble spark is used it is necessary to use a dye with alcohol as solvent. By

FIG. 5.

FIG. 6.

using a mixture of alcohol and water with a dye soluble in either, a satisfactory solution is obtained for any size spark hole and any specific duration of exposure to the solution. This system permitted the use of primary voltages as low as 50 volts instead of the 220 volts necessary with the Aberdeen chronograph paper.

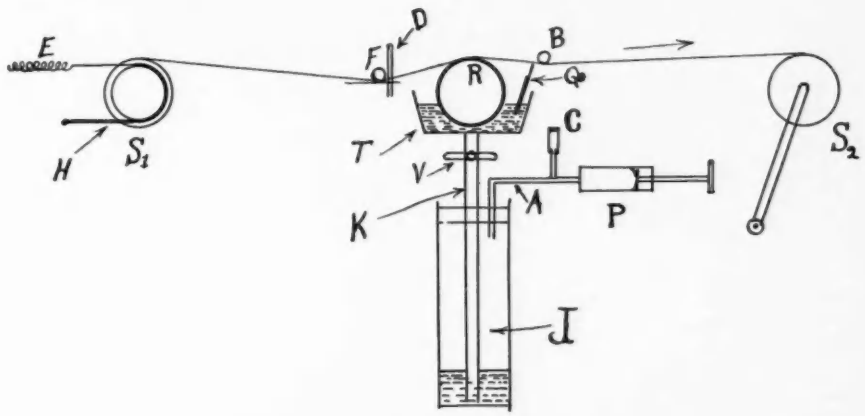


FIG. 7. *Developing machine.*

Figure 7 shows a cross sectional diagram of the developing machine. The paper passes over the roller (*R*) which is rotating in the developing solution (solution of water, alcohol and dye). The paper then passes

over the rubber squeegee (Q) which scrapes off all the surplus developing solution. Since a paper with waterproof coating is used, it is not necessary to dry the paper after leaving the squeegee but it may be immediately wound up on the spool (S_2). The spool (S_2) is provided with a crank which is operated by hand. There is a friction brake on the spool (S_1) from which the paper is to be pulled. This regulates the amount of tension on the paper and keeps it constant. The idler rod (F) is floated on the paper and helps smooth out irregularities in the brake tension.

For convenience in inserting the developing solution into the tray (T) the solution is placed in the bottle (J). A tube (K) having a valve (V) extends from the bottom of the bottle into the tray (T). There is a pump (P) by which air may be pumped into the bottle through the tube (A). This forces the developer up into the tray (T). The valve (V) is then closed.

When the liquid is to be removed from the tray, the valve (V) and the port (C) are opened, permitting the air to escape and the developer to run back into the bottle (J).

A great deal of data was obtained in the Laboratory to determine the accuracy of reproducing hammer velocity with a constant pressure in the wind chest of an Ampico. It was found that the hammer velocity could be reproduced with an accuracy of better than five parts in one thousand in playing a single note.

Considerable data was obtained in order to find the effect of measuring the velocity by changing the distance between the contact points. It is interesting to note that as long as the first contact was not made prior to the time when the hammer was about 12 mm. from the string, that the velocity measurements remained the same, regardless of the distance between the contacts. In other words, if the first contact is made 11 mm. before the hammer strikes the string and the second contact is made 1 mm. before the hammer strikes the string, the velocity is found to be the same as when the first contact is placed 6 mm. before the hammer strikes the string and the second contact 1 mm. before the hammer strikes the string. All this was done in order to show that the velocity of the hammer does not change materially during the latter portion of its travel. In other words, after the hammer reaches a point about 12 mm. from the string the velocity remains practically constant until it strikes the string.

Figure 8 shows the manner in which the contacts are made by the motion of the hammer. The contactor (C_0), which is a light silver bar,

is attached rigidly to the shank (L). This contactor is connected to the bus bar (G) by means of a small flexible copper wire. There are two silver wires, C_1 and C_2, which make contact with the contactor (C_0). As the contactor (C_0) touches the silver wire (C_1), the primary circuit is closed and a secondary spark occurs between the rotor and the segment to which C_1 is connected. In like manner, when C_0 contacts with C_2 another spark occurs between the rotor and the segment which is connected to C_2. The bars (D) are the distributors to which the contacts (C_1) and (C_2) are connected. The contact (C_2) has a definite down stop located so that the contactor (C_0) engages it about 1 mm. before the hammer strikes the string (S). The contact (C_1) is adjustable. Its

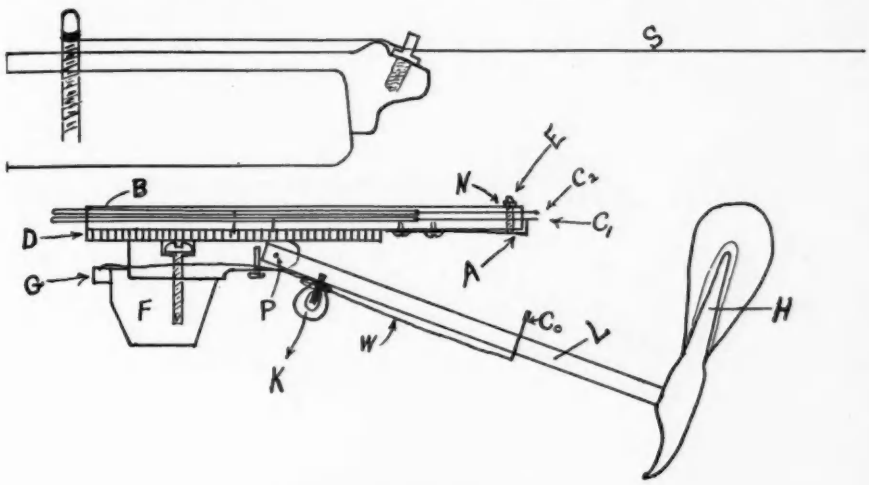


FIG. 8. *Recording contact system.*

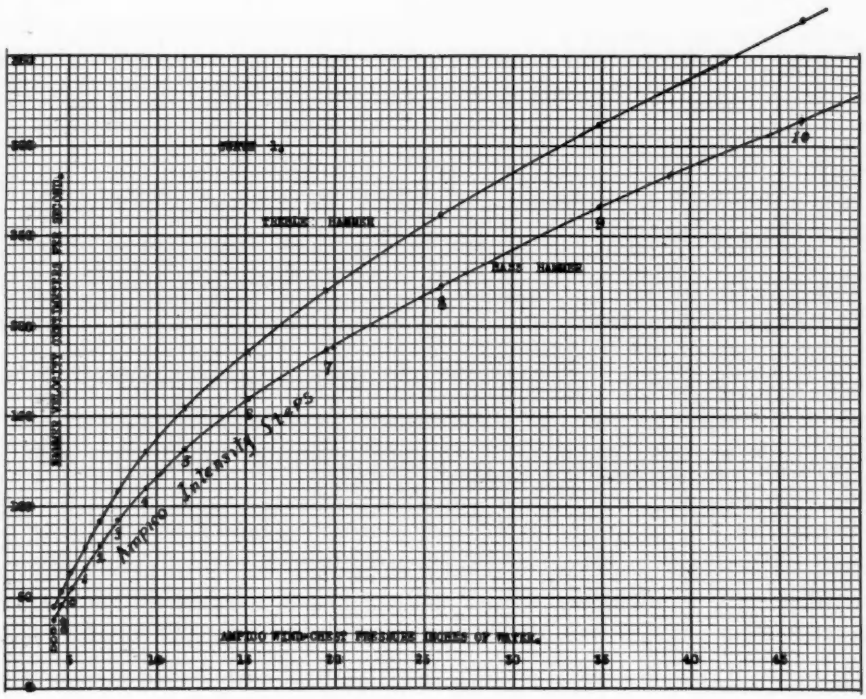
down stop being the spring (A) which is controlled by the screw (E) and locknut (N). This contact is made about 4 mm. before the hammer strikes the string.

The distance between C_2 and C_1 is adjusted by means of a micrometer. It is interesting to note that after eighteen months daily use in the Recording Department, the contact distances for all the hammers were measured and with the exception of three notes were found to be correct to better than 1%.

Curve #1 shows the velocity of a bass and treble hammer respectively for various Ampico wind chest pressures. The dots on each curve represent the Ampico steps of intensity. The velocity of the hammer and also the wind chest pressure for each of these intensities may be

read from the graph. The velocity of the treble hammer is appreciably higher than the velocity of the bass hammer, however, the mass of the bass hammer is greater, so that the energy remains approximately the same. It is the intention of piano builders to keep the loudness of the bass and treble portion of the piano well balanced.

Curve #2 shows the steps of intensity of the Ampico plotted against the logarithm of the square of the hammer velocity. Here it is seen that the slopes of the curves for the two hammers are almost identical. The increase in energy can be determined directly from the logarithms

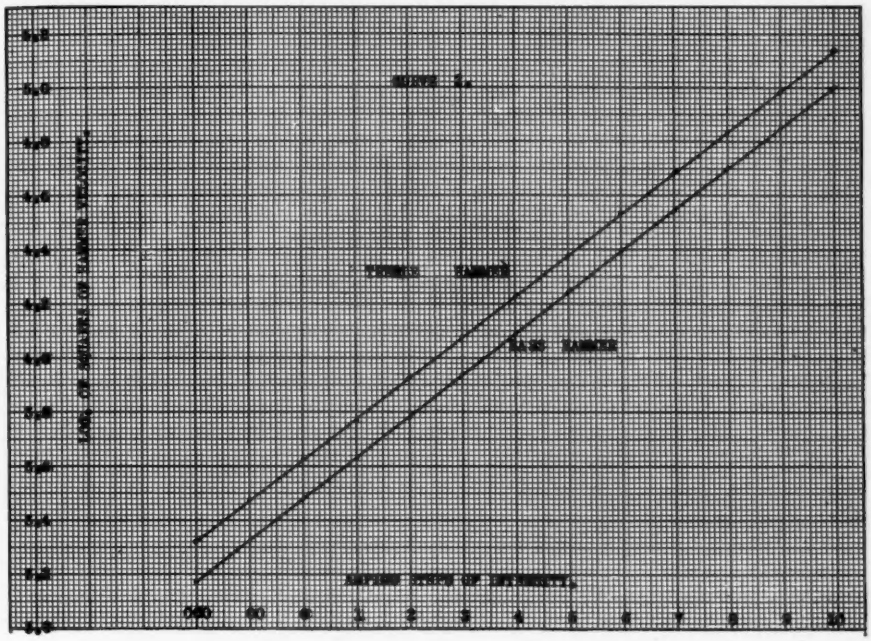


and shows that in each hammer there is an increase in loudness corresponding to 18.2 decibells.

Curve #3 shows Ampico steps of intensity plotted against sensation units or decibells. The only assumption that is made here is that the sensation level for the lowest Ampico step for a single note is 30 decibells (Sensation units). The loudest Ampico step brings the intensity level up to 48.2 decibells. This curve represents the values for both bass and treble hammers. This is on the assumption that the intensity level in each hammer starts at 30 decibells. This is an arbi-

trary value but the value is the same for both hammers because the Ampico, before leaving the factory, is adjusted so that, for the lowest intensity, equal degrees of loudness are obtained from all notes. The value of 30 decibells is arbitrary, being chosen according to the best judgment available.

While this intensity range of 18 decibells may seem small, it should be remembered that this range is for a single note. When an artist is playing fortissimo and using the sustaining pedal, it is possible to build up an intensity perhaps twenty times as great, which would add an

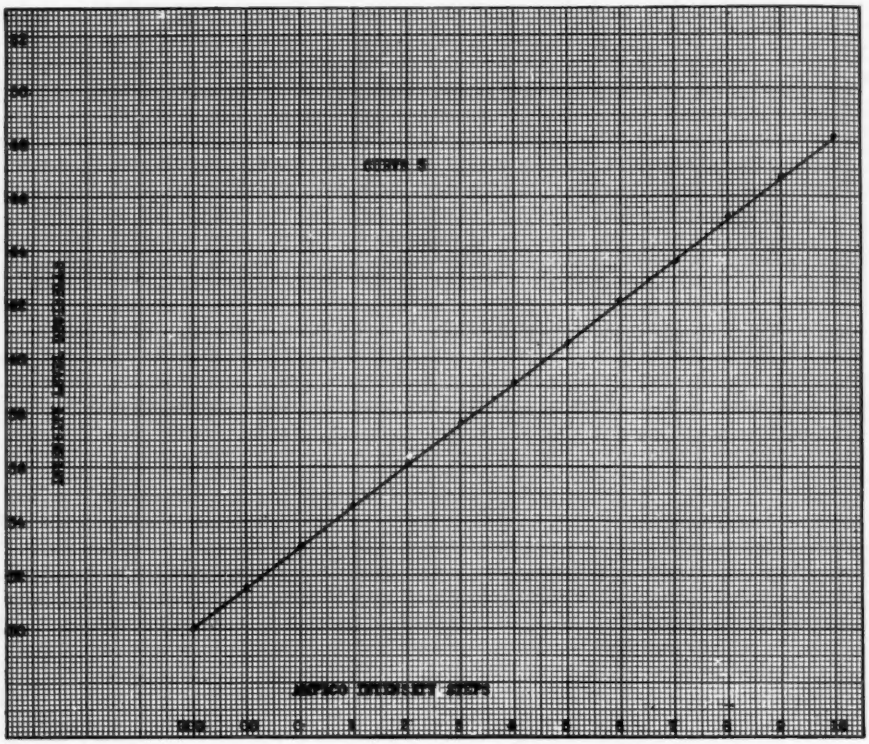


intensity of about 13 more decibells to the range of the piano, giving a total of about 31 decibells.

It has been shown that the velocity of a bass hammer is less than the velocity of a treble hammer for the same level of loudness. This change in hammer velocity is not an abrupt one but takes place gradually from one end of the piano to the other. This is to be expected since the hammers are all different in size. They are moulded in a tapering form and then later sawed apart.

In order to avoid the necessity of using a different measuring scale for each hammer, the contact distances in the bass portion of the re-

cording piano are made smaller than those in the treble. Each hammer has a different contact distance. The total variation from note #1 to note #88 is about .030 of an inch. This gives approximately .0003″ difference in contact distance for adjacent notes.



It is hoped that chronographs of this type may be useful for other purposes. A great number of records may be made simultaneously on stationary paper. The paper may, however, be moved along slowly. It is only necessary to move the paper about 1/32″, after one record has been made in order to obtain another record from the same segment.

It is also hoped that the recording paper principle which has been described may be useful to others.

METHODS AND APPARATUS FOR MEASURING THE NOISE AUDIOGRAM

By Rogers H. Galt
Bell Telephone Laboratories

Among the characteristics of a noise, one of the most significant is its deafening effect upon the human ear. This property is the basis of the engineering definition of a noise as an extraneous sound which interferes with the reception of other sounds. In common experience, the hearing of speech or of music is continually subjected to interference by such noises as the din of street traffic or the shuffling of feet and banging of chairs in a room. The normal method of overcoming such sounds in conversation is for the speaker to talk more loudly in order that the listener may hear him; in other words, the speaker must act as if the listener were temporarily suffering from partial deafness, due to the noise.

A measure may be assigned to the deafness produced by a noise in fundamentally the same manner as that in which deafness due to other causes is measured, namely, by observation of the threshold of audibility. Threshold studies of the normal ear have determined for each frequency the intensity of the acoustic wave which is just audible in a quiet place. To a partially deaf ear, over all or part of the frequency range a tone of normal threshold intensity will be inaudible, an abnormally great intensity being required if the tone is to be just audible. The difference between the threshold intensities for the normal and for the deaf ear is taken as a measure of the loss of hearing at the particular frequency involved. If the loss of hearing is plotted against frequency, the resulting curve is the audiogram of the deaf ear. Such a curve showing the loss of hearing at each frequency due to a noise is termed a noise audiogram.

The usual method of obtaining an audiogram of an ear, by threshold tests involving single frequency test tones, may in certain cases be applied to measurements of the deafness due to a noise. There is an important difference between the two types of measurement, however, which renders it necessary in the case of many noises to use complex testing waves. Since the deafness due to a noise largely disappears as soon as the noise has ceased, the measurements must be made while the noise wave is affecting the ear; if this wave contains a prominent

component nearly equal in frequency to a single frequency testing tone, beats between the two simultaneously applied waves occur which invalidate the threshold observation.[1] In the use of an adequate complex testing wave, the beats associated with any one component become of negligible significance.

Certain general requirements must be satisfied by complex testing waves to be employed in a device for measuring the deafness due to noise. The waves must be producible with simple and portable apparatus. The testing sounds must be distinctive in type, to insure recognition near threshold. Each wave should contain components limited to a definite frequency interval, the various intervals being so distributed as to cover the most important part of the range of audible tones. The total number of components in any testing wave must be great enough to render insignificant, in the totality of sound received by the ear, any beating phenomena between individual components of the testing and the noise waves.

These requirements are satisfied by a testing wave of the warble type. In such a sound the pitch varies continuously from a lower to an upper limit and back again. Analytically, the periodic wave known as a warble may be shown to consist for practical purposes of discrete steady single frequency components lying between the band limits and separated by successive frequency intervals each equal in magnitude to the warble frequency. A warble may therefore be located in a desired frequency interval, with chosen spacing of components. The warbling sound is sufficiently unusual in type to be recognizable in the presence of almost any noise. The phonograph affords a convenient and portable mechanism for generating the sounds.

In the apparatus about to be described, the testing waves are warbles of a fundamental periodicity of about six cycles per second. Too great a departure from this rate of warble involves a sacrifice of some of the desired wave characteristics. For instance, if only one warble is executed per second, the tone remains nearly fixed in pitch long enough to build up disturbing beating effects with prominent components of some noises; and with too fast a warble, say fifteen periods per second, the sensation of sweeping through a range of pitch is submerged in the less distinctive effect of a fluttering complex tone.

As the primary generator of these testing waves, it was found convenient to use an oscillator of the heterodyne type with one oscillating

[1] Wegel and Lane, Physical Review, second series, vol. 23, No. 2, Feb. 1924.

capacity fixed and the other continuously varying at the desired rate of six periods per second. Adjustment of the oscillating elements permitted ready control of the warbling band limits. By means of electrical networks, the relative amplitudes of the various components of each testing wave were so adjusted that in the final apparatus, when the warbling sound is applied to the normal ear, all components of a wave are at the threshold level for the same setting of the volume control.

The heterodyne oscillator, with the associated amplifier, batteries, filters and other networks was regarded as too cumbersome for incorporation into the final noise measuring set. Consequently each warbling test wave produced by the oscillator was recorded phonographically; the resulting records, together with a phonograph for reproducing the waves electrically and other apparatus for controlling their volume and for applying each wave and the noise together to the ear, constitute the portable device under description.[2]

The incorporation into the set of so flexible an element as the phonograph renders possible a wide variation in testing waves by the simple process of changing records. This is of particular value in adjusting the fineness of the frequency analysis to be afforded. In one arrangement the frequency range from 160 to 5400 cycles per second is divided into six intervals, one warbling test wave covering each interval; the six waves are recorded, three on each side of a two-faced phonograph record. In another arrangement, the range from 250 to 5600 cycles per second is divided into three intervals, the three waves being recorded on one face of a record. Other arrangements are of course equally possible.

The frequency limits of the various testing waves actually recorded are given in the following table:

	Band Limits In Cycles per Second
Six-Band Record	160– 240
	260– 500
	500–1000
	1000–2000
	2000–3000
	3100–5400
Three-Band Record	250– 750
	750–1500
	1500–5600

[2] "Speech and Hearing", H. Fletcher p. 105.

The complete apparatus which employs these waves in noise measurements is contained in two boxes weighing together about forty-eight pounds, each furnished with a handle for carrying in suit-case style. In Figure 1 is shown schematically the path by which the testing wave passes from the phonographic source to the ear. The Western Electric Type 101-A electromagnetic reproducer with a particular type of extra loud needle reproduces each testing wave from the phonograph record; the electrical wave thus generated traverses an adjustable network of resistance elements termed an attenuator which conveniently and distortionlessly controls the voltage applied to a Western Electric iron clamped, paper damped receiver. Between the reproducer and the attenuator, an electric wave filter is employed with the low frequency testing waves, passing all frequencies up to 1000 cycles per second, but suppressing all the upper range in which are located the most disturbing components of phonographic surface noise, or scratch. As such

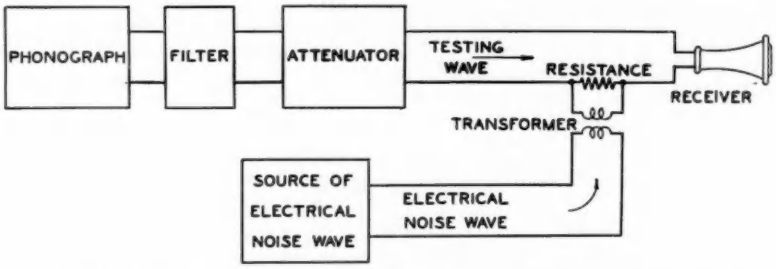


Fig. 1. *Apparatus For Measuring The Audiogram of An Electrical Noise.*

filtering is not required for the higher pitched testing waves, which moreover lie within the region suppressed, a switch is provided by means of which the filter may be cut out.

The path of the testing waves, from the phonograph record through reproducer, filter when employed, attenuator and receiver to the ear must at some point meet the path of the noise leading toward the ear. If the noise is available in the form of an electrical disturbance, such as the electrical wave producing a hum in a telephone line, its path is connected to that of the testing wave in the electircal circuit immediately before the receiver in some such manner as that indicated in Figure 1. If the noise is available in acoustical form, such as room noise, its path joins that of the testing wave after the latter has become an acoustical wave, between the receiver diaphragm and the ear, as shown in Figure 2. To accomplish the union of the two acoustical paths, one from the

receiver diaphragm to the ear, the other from the room space to the ear, a special receiver cap termed an off-set cap has been designed. This cap has the customary central hole, to permit sound waves to pass from the receiver diaphragm to the ear; in addition slots are provided, large enough to admit the room noise with negligible distortion, yet small enough and so disposed as to allow a minimum of escape of diaphragm sounds into the room.

The procedure in making a measurement is to allow the noise and one of the warbling test waves to reach the ear simultaneously, and to alter the intensity of the warble by means of the attenuator of Figures 1 or 2 until it is just audible in the presence of the noise. The dial setting of the attenuator corresponding to this adjustment is observed, and from it is

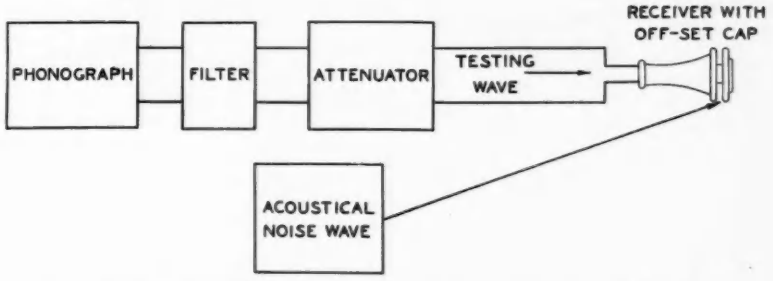


FIG. 2. *Apparatus For Measuring The Audiogram of An Acoustical Noise.*

subtracted the setting previously found for the same testing wave under noiseless conditions. The difference between the two attenuator settings gives in decibels the change produced by the noise in the threshold of hearing for the frequency interval covered by the particular testing wave employed. Similarly, threshold observations are made in succession upon the other testing waves.

An upper limit to the deafening effect measurable with the apparatus in any frequency band is set by the acoustic power delivered to the ear when the corresponding testing wave is being reproduced with the attenuator adjusted for zero loss. If the noise is loud enough to mask the testing wave at its greatest intensity, a threshold determination is impossible. Although the measurable limit may be increased by incorporating a suitable amplifier in the apparatus, this has not yet been done owing to considerations of portability and of the generally satisfactory range of the present outfit. The set as constructed has a range sufficient to include practically any electrical noise which occurs on tele-

phone lines and almost any room noise up to those levels which render conversation extremely difficult.

Examples of the results of measurements made with the apparatus are given in Figures 3-6. In Figure 3, three audiograms illustrate the six band analysis of the noise deafening under three quite different conditions. Audiogram No. 1 indicates the condition in a booth having walls lined with a sound absorbing material, the door of the booth being open to the adjoining office, the windows of which were closed. This curve describes a distinctly quiet office condition. Audiogram No. 2 shows the deafening observed in a large room without absorbent padding with windows open and numerous people moving about and conversing. Audiogram No. 3, indicating an especially great loss of hearing at the

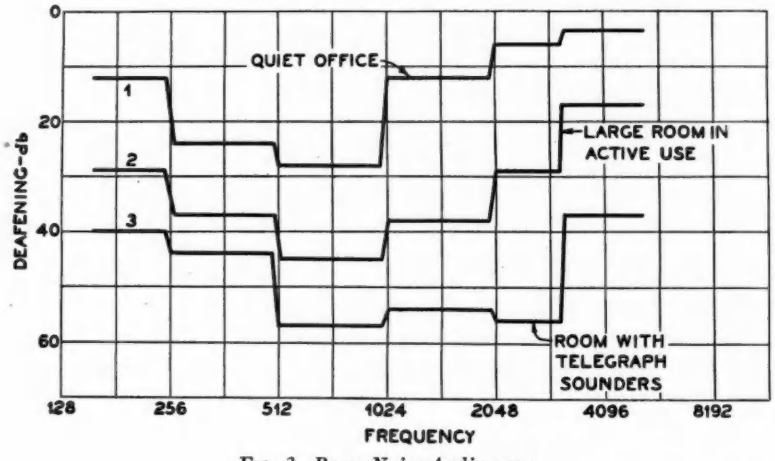


Fig. 3. *Room Noise Audiograms.*

high frequencies, was obtained in a large unpadded room, with windows open and people moving and talking, and with a number of telegraph sounders operating.

Figure 4 illustrates the deafening as analyzed by the three band arrangement. Audiograms No. 1 and No. 2 were taken in an ordinary tenth floor office with windows respectively closed and open; audiogram No. 3 applies to a large room with windows open upon a noisy street adjacent to wharves, and with about twenty typewriters in operation; audiogram No. 4 was obtained in a large wood shop where saws and other noisy apparatus were operating.

In Figure 5 are shown certain observations made at the time of the homecoming reception on lower Broadway, New York City, to Colonel

Charles Lindbergh following his transatlantic flight. The observers were located at an open window on the north side of the fifth floor of a

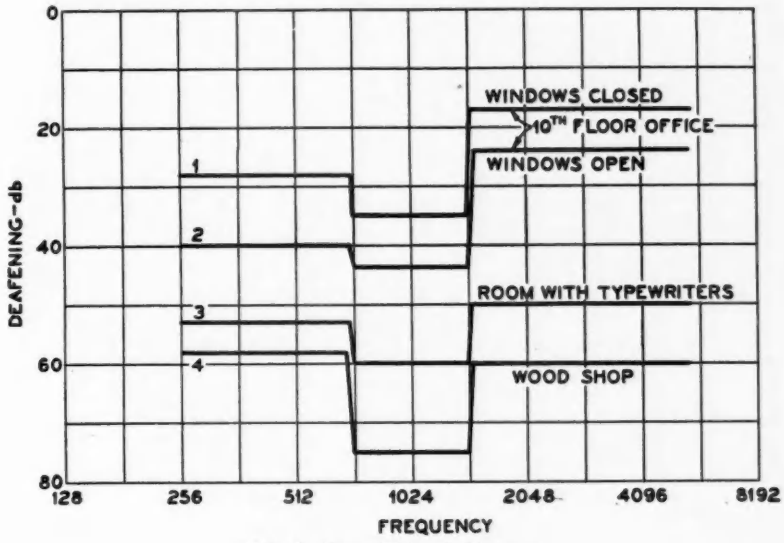


FIG. 4. *Room Noise Audiograms.*

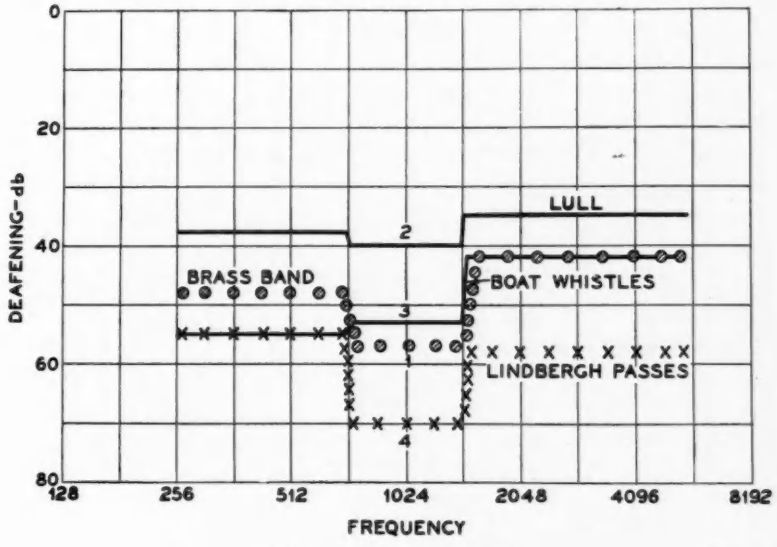


FIG. 5. *Noise Audiograms.*

building fronting on the West side of Broadway. The window was about one hundred feet west of the building front, and about fifty feet

above the level of the Broadway sidewalk. Audiogram No. 1 shows the observed deafening effect when a brass band, with drums, was passing; audiogram No. 2 was taken in a subsequent lull in the street noise, which seemed by contrast extremely quiet; audiogram No. 3 shows the effect, very intense at low frequencies, upon the observers' ears when the aviator's arrival at the Battery, nearly three-fourths of a mile distant, was proclaimed by steam-boat whistles; audiogram No 4 gives the deafening due to the crowd's welcome as Colonel Lindbergh passed. The level at the center of the Broadway crowd was of course higher than that at the window where the observers were located; high enough, in fact, to mask the sound of a brass band not many yards distant. Perhaps in the future, those concerned in feeling the pulse of popular enthusiasm

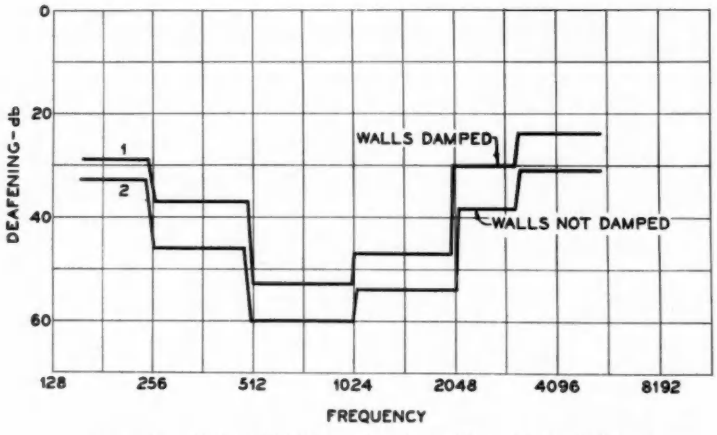


Fig. 6. *Effect of Wall Damping Upon Room Noise Audiogram.*

may use some such methods of observation to gauge the fervor of the crowd.

Figure 6 contains the results of tests upon a room especially constructed to study the effects of changes in wall absorption. By means of removable panels of an absorbent material, the reverberation time for a tone in the range 100 to 3000 cycles per second could be altered from about 5 seconds to about 1 second. Noise at a fairly constant level was admitted to this room through two slotted windows, about a foot apart and each two feet square, connecting with a smaller room containing a phonograph. The phonograph reproduced a record of the noise of a large room adjacent to a noisy street and containing numerous persons moving about, talking and operating typewriters. The noise level in the testing room, due to the phonograph in the smaller room, was about

that characteristic of the place where the phonographic record had been made. The construction of the two rooms was such that external sounds contributed only a negligible amount to the room noise when the phonograph was operating. In Figure 6, audiogram No. 1 indicates the deafening effect of the noise when the damping panels covered the walls of the testing room; audiogram No. 2 shows the deafening effect with the panels absent. For the noise level employed the result of introducing the absorbent material was to cause an average reduction in the deafening effect over the frequency range examined by about 7 decibels. This result agrees quite closely with the effect which would be expected on the basis of the reduction of energy density of the noise waves by an amount of absorption responsible for a five-fold decrease in reverberation time.

A measurement of the amount of deafening due to a line or a room noise is affected by several variable factors, of which some are due to the type of measurement itself and others to the apparatus involved. The former causes of variation, which are more important than the latter with the present apparatus, are two-fold. In the first place, the noise level itself is rarely a constant quantity, but in the case of room noises usually changes with time over a range of ten decibels or more, and in many instances varies by such an amount with position in the room. Secondly, auditory and psychological differences between different observers having normal hearing, or between the responses of the same observer at different times, may introduce variations as great as five to ten decibels in the threshold setting of the attenuator in the presence of the noise.

The apparatus contributes distinctly smaller variations. The iron-clamped type of receiver is constant in its response to within plus or minus one-half decibel. The change in level of the reproducer output due to wear of the phonograph record is negligible during the life of the record, which is limited to fifty or sixty playings in the high frequency band due to the subsequent development of excessive scratch. The effect of needle variations is not more than one decibel ordinarily. Finally, the reproducer contributes variations which will not exceed three decibels provided standard maintenance rules are observed.

The reproducibility of measurements with the present apparatus is therefore largely dependent upon the constancy of the noise level under test and upon the observer's hearing. A single observation by one observer will generally indicate the magnitude of the deafening effect of a constant noise to within five or ten decibels of the mean of several ob-

servers using different measuring sets. The number of observers and of observations to be made in any particular case will depend upon the specific conditions and requirements.

The manner in which an observation upon ordinary room noise is made is of considerable importance. Since the observer is endeavoring to discover the deafening effect of the noise, recognition must be given to the variability of such an effect in ordinary noises. The procedure is not that of altering the test wave level extremely slowly until a highly accurate threshold level is found, but rather that of repeatedly and rather rapidly determining the threshold, each observation being noted. These observations will then afford limits of variation of the deafening effect, or they may, for each frequency band, be consolidated into a single measure which will indicate in a broad way the deafening effect.

In calibrating the set for zero noise, which is done in a sound-proof booth, it is necessary to increase the number of observers and of observations considerably beyond those required for the threshold tests with noise present. Whereas a single observer, with two or three observations upon each band, can obtain a reasonably satisfactory threshold setting in the presence of a noise, it is advisable to employ four or more observers, with several observations each, to determine the threshold setting for the condition of no noise. This necessity arises from the relatively low precision of the zero noise threshold observations. As long, however, as the apparatus is handled with ordinary care and the current output of the reproducer into a resistance of 500 ohms is tested occasionally and found within prescribed limits, the same calibration for zero noise will hold.

Each record face contains three bands, separated by two ten second silent intervals. The first two bands on each record are about 45 seconds in duration; the third about 90 seconds. As the time required for a threshold observation in the presence of a steady noise is quite short, it is generally possible to repeat an observation several times before the testing wave changes, if the noise is fairly steady, and to form some idea of the limits if the noise is changing rapidly. If the noise possesses the intermittent character of hammer blows, anything approaching accuracy is difficult to attain in the measurements. However, it is often possible to find two attenuator settings such that for one, the reference sound is never masked by the noise, and for the other, the reference sound is definitely not heard during the noise peaks.

Although the apparatus described has been found to be a satisfactory general method of measuring the loss of hearing due to a noise,

other methods may also be employed within their proper limits. For instance, the audiometer used in determining the audiograms of ordinary deafness, employing single frequency test tones, will afford an analysis of the deafness due to noise, provided, as already pointed out, the character of the noise is such as to introduce no significant phenomena of beats. In order to explore the spectrum as completely with a single frequency source as with a complex wave source, it is probable that a larger number of testing waves would be required.

Among other types of noise measurements one only will be mentioned for comparison, employing a single complex testing wave usually generated by a buzzer of fundamental frequency about 150 cycles per second. When a threshold test with such a tone is made in the presence of a noise, a single observation is obtained which may yield information in some particular part of the spectrum equivalent to that obtained from one of the warbling test waves of the present apparatus. No analysis of the deafening effect is afforded, however, and not only is the wide spacing of components, 150 cycles apart, liable to lead to the trouble of beats, but the tone itself is often less easily recognized than a warble, increasing the uncertainty of the test.

Although a single observation upon a noise yields results of value in many circumstances, a frequency analysis is more significant in the problems of communication engineering because of the unequal importance of the various frequency intervals in speech. Moreover, in physiological and psychological studies of the effects of noise, which are only now beginning to arouse general interest, and in investigations of noise prevention and reduction, an increasing importance may be assigned to frequency analysis. The principal merits of the present apparatus are therefore that it is a portable device which affords a means of obtaining rapidly the audiogram of a large variety and range of ordinary noises, with adjustable analytical power.

In conclusion, the author desires to acknowledge the contributions of Dr. H. Fletcher and Dr. J. C. Steinberg to the fundamental ideas involved and the general plan of the present apparatus.

April 23, 1929.

PSYCHOLOGICAL MEASUREMENTS OF ANNOYANCE AS RELATED TO PITCH AND LOUDNESS

By Donald A. Laird and Kenneth Coye
Colgate University Psychological Laboratory
Hamilton, N. Y.

In some acoustical installations it is observed that a small reduction in loudness is accompanied by a distinct feeling of relief, while under other conditions a much greater reduction in loudness is not followed by a proportionate feeling of relief. In consequence of these widely observed reactions many acoustical workers have suspected that the annoyance experienced by workers as a result of noise was a function of pitch as well as of intensity.

The outstanding mental reactions of which workers are directly conscious as a consequence of noise are in the realm of the feelings,—mild emotional-like states almost invariably associated with sensory experiences. Titchener has characterized this psychological basis of annoyance as "an elementary affective process, a feeling-element which in our minds is coordinate with sensation and distinguishable from it, but which is nevertheless akin to sensation and is derived from the same source." Although this annoyance element may be given a personal variation as a result of associated states in previous experiences, it is generally recognized that it is usually intrinsically dependent upon conditions in the sense organs and thalmus in the mid-brain which do not vary markedly from individual to individual.

The annoyance reaction to an auditory stimulus is similar to that experienced by rubbing one's fingers over coarse sand-paper, or in biting into a piece of grit in candy, or in hearing peanuts shelled during an overture. It carries with it an unpleasant feeling-tone, a tinge of restlessness and tension. When stimuli differing in pitch are used it is relatively easy to identify this experience; when stimuli varying only in loudness are used trained observers are necessary. So that the data would have greater accuracy trained observers were used for these experiments. All observers were young men who had had considerable experience in similar observations.*

*The following trained observers served throughout the experiments: W. G. King, Herbert C. Bush, W. L. Webb, J. C. Anderson, Robert Houston, F. K. Berrien, E. Stanley Copeland, G. W. Kellogg, Everett Holt, Walter Smith, James Lewis, and Ralph Snyder. Records from the authors are not incorporated in the report due to possible bias.

In acoustical quieting it is undoubtedly reducing this state of annoyance that brings the experience of blessed relief to the workers. This is apparently in addition to a reduction of the fear reaction for the annoyance may still be present to a degree with a loudness too low to cause a demonstrable fear reaction.

Stimuli for the experiments were obtained directly from the 2A audiometer. All observations were made in the cubical test chamber in the Colgate laboratory; this is ten feet in each dimension and ceiling and side walls are lined with type BB Acousti-Celotex. Artifical lighting and forced ventilation is used. Outside noises were not a disturbing factor.

Determinations by Paired Comparison Method

The first data were obtained by the well-known method of paired comparison. The audiometer sensation level of the eight pitches was kept at 50 t. u. Each pitch was presented to the subjects paired with each of the other seven pitches, in turn, and the subjects reported which of each pair was the more annoying. Twenty-eight pairs of pitch combinations were made, following the scheme below, in which the numeral in the intersections are ordinals indicating the pitches which were compared first, second, etc:

Scheme Followed in Presenting Pairs of Pitches to Subjects

Pitch	64	128	256	512	1024	2048	4096	8192
64		1	8	14	19	23	26	28
128			2	9	15	20	24	27
256				3	10	16	21	25
512					4	11	17	22
1024						5	12	18
2048							6	13
4096								7
8192								

There was not a single pitch but which in some combination was annoying to some observer. The three pitches which were least annoying were 256, 512, and 1024 cycles; out of 28 paired comparisons each of these was indicated as the more annoying on an average of 1.5 times. By the method used it would have been possible for these pitches to have received a report of zero rather than 1.5, so it must be inferred that at this sensation level some observable annoyance will be caused by any of the pitches used.

The higher pitches are distinctly more anoying than these three more neutral tones, but, unfortunately, we cannot just yet say how much more annoying they were. It has been generally believed that the high pitches were the more annoying, but the status of the low pitches has

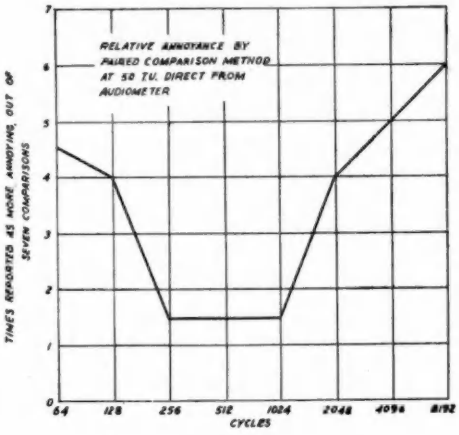


Chart I

been a matter of question. Our data show clearly in Chart I that 128 and 64 cycles are as annoying as 2048. The most annoying is 8192 cycles, which was reported as the more annoying in an average of six out of each seven frequencies with which it was paired.

Determinations by Equal Annoyance Method

Data of more practical and theoretical value were obtained by making determinations of equal annoyance. As a standard for comparison 256 cycles at intervals of 10 t. u. on the audiometer scale was used. The 256 intensity was sounded until the observer signalled that he had the annoyance of that intensity clearly grasped, following which another pitch was immediately sounded and the audiometer dials adjusted until the observer pronounced the annoyance to be the same as the first, or standard, annoyance. A second check observation was made before the attenuator figure was recorded; checks were also run on later days.

Records were obtained for the sensation levels of each of the audiometer pitches which yielded the annoyance equivalents of the 8 intensities of 256 cycles which were used as standards. Chart II shows the averaged records from the 14 observers. There were some individual differences, some observers running consistently higher than others, but

the relative positions of the lines were practically the same for all and but little smoothing was needed in charting.

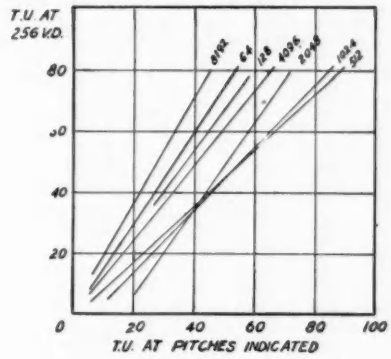


CHART II

It is apparent that in absolute values there is more difference in the annoyance of the various pitches at the high intensities than at the low intensities. Thus 512 cycles at 89 t. u. and 8192 cycles at 44 t. u. are of equivalent annoyance level to 256 cycles at 80 t. u., a difference be-

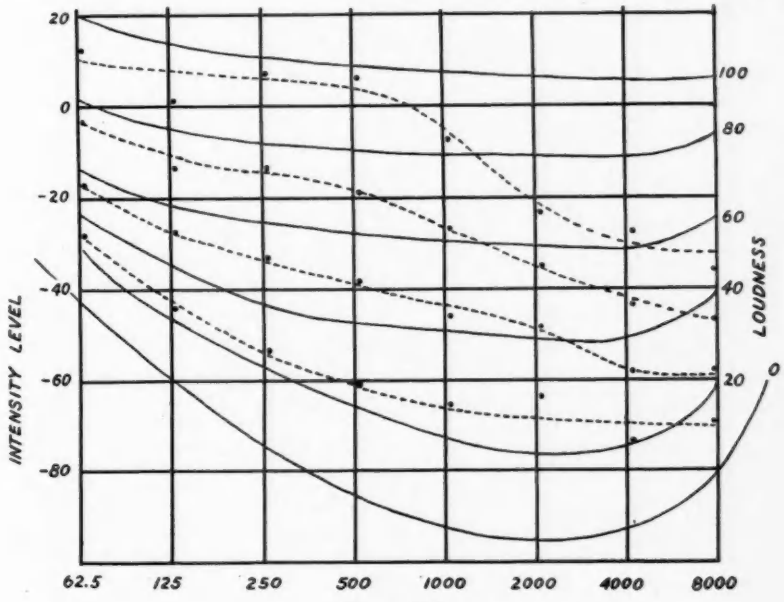


CHART III

tween 512 and 8192 of 45 t. u. At the annoyance equivalent to 256 cycles at 20 t. u. we find 8192 cycles at 9 t. u. and 2048 cycles at 31 t. u. representing the extremes,—a difference of only 22 t. u.

These results corroborate those shown in Chart I, the high and the low frequencies being more annoying than 512 and 1024 cycles. This is marked above the 40 t. u. intensity while below 20 t. u. 2048 cycles becomes the least annoying.

When the data of Chart II are interpolated into relation to loudness in Chart III we find that for the low intensities annoyance is approximately proportionate to loudness for the different frequencies. At the highest annoyance level charted, where 256 cycles at 80 t. u. was the arbitrary standard, we find the pitches below 500 following the loudness curve, while pitches above 500 are equally annoying at lower loudness levels. This tendency is noticeable from the annoyance value of 256 cycles at 40 t. u. and upward, but becomes more marked at the higher loudnesses.

That annoyance is different from loudness is apparent from this chart. That high pitches are intrinsically more annoying than low or medium pitches is also evident. Annoyance also appears to be dependent upon the loudness of the stimulus, as is shown by the records below 500 cycles but the tendencies with reference to pitch are marked.

General Discussion

It has not been possible for us to discover as yet any method for measuring absolute annoyance; this is a pressing and important problem. The observers are of the opinion that there is a close relationship between loudness and annoyance, but whether this is a straight line or a curvelinear relationship can only be conjectured. If this could be definitely established we would then have a basis for the determination as annoyance from a definite zero point.

It is of considerable theoretical interest that those pitches which man himself makes in speech are the least annoying to him. The annoyance of certain tenor and soprano voices where higher pitches occur is in line with our findings. The low annoyance values of the more common speech sounds may represent a biological adaptation.

It has been suggested that the relatively increased annoyance value shown for the low cycles in Chart I may be due to association of these low tones with physical vibrations. On a similar basis the increase in annoyance as the threshold of feeling is approached might be misleadingly conditioned.

Most commercial sound absorbing materials have their highest efficiency between 256 and 2048 cycles, while it appears to be the pitches below and above these that are the most annoying and should obvi-

ously be reduced most. Practically this may be offset, however, since audiograms of street noises and typewriter noises given by Fletcher indicate in general their frequencies to follow the more usual absorption curves.

The data reported may be found of value in determining whether or not acoustical installations should be made in work places where the actual noise intensity is not great. When the noise pattern in a room contains 8192 cycles at only 40 t. u. it is apparent from Chart II that quieting is needed to offset annoyance as badly as though the room had 256 cycles at 72 t. u. The later intensity would sound distinctly loud to the ear while the less loud 8192 cycles is as annoying and correction is plainly as needed. Given the right pitches, a room can be relatively quiet but still annoying. Our data give no basis for assuming that silence is annoying unless a curvelinear relationship is established between loudness and annoyance.

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THE JOURNAL
of the Acoustical Society of America

Volume I Number 2, Part 1

JANUARY · 1930

PROGRAM OF THE SECOND MEETING OF THE ACOUSTICAL SOCIETY OF AMERICA

DECEMBER 13 AND 14, 1929

All meetings held in Lincoln Hall,
Northwestern University School of Law
357 E. Chicago Ave., Chicago, Illinois

FRIDAY, DECEMBER 13

9:30 A.M.

1. **Address of Welcome by President;** Business (20 minutes).
2. **Symposium on Sound-Insulation**—Measurements and Practical Types of Construction.
   (a) V. L. CHRISLER, *Bureau of Standards* (25 minutes).
   (b) PAUL E. SABINE, *Riverbank Laboratories* (25 minutes).
   (c) F. R. WATSON, *University of Illinois* (25 minutes).
   (d) ROBERT L. DAVISON, *The Architectural Record* (25 minutes).

   Discussion—

   WALLACE WATERFALL, *The Celotex Company.*
   R. V. PARSONS, *Johns-Manville Corporation.*

   Discussion by members.

12:30 P.M. to 2:00 P.M.

Luncheon

FRIDAY, DECEMBER 13
2:00 P.M.

1. Reverberation Time in "Dead" Rooms, CARL F. EYRING, *Bell Telephone Laboratories* (20 minutes).

An analysis based on the assumption that image sources may replace the walls of a room in calculating the rate of decay of sound intensity after the sound source is cut off gives the more general reverberation time equation

$$T=\frac{.05V}{-S\log_e(1-a_a)},$$

where T is reverberation time, V the volume of the room, S the surface of the room, and a_a the average coefficient of absorption of the wall surface. In the past Sabine's formula,

$$T=\frac{.05V}{Sa_a},$$

has been used to calculate reverberation time, but for rather large values of a_a, that is, for "dead" rooms this equation gives too long a time of reverberation, or in some cases, if the ime is measured, it gives a value of a_a greater than unity. This we found to be the case in the Sound Stage, Sound Picture Laboratory, Bell Telephone Laboratories, Inc. Reverberation time measurements made in this room support the new formula, however. Sabine's formula turns out to be the special case of the more general equation and is to be used when a_a is small, that is, for "live" rooms, but not for "dead" rooms.

The new formula is of importance in the talking picture industry for it indicates that a "dead" room may be obtained by the use of less absorbing material than that calculated by the old formula.

October 17, 1929.

2. A Chronographic Method of Measuring Reverberation Time, E. C. WENTE and E. H. BEDELL, *Bell Telephone Laboratories* (20 minutes).

A description is given of apparatus which measures the rate of decay of sound density in a room. A loud speaker, energized from an oscillator, is preferably used as the source of sound. Current is sent through the speaker until the sound in the room has reached a steady state, whereupon the speaker is disconnected from the oscillator and the time required for the energy density in the room to decay to a definite value is recorded by the apparatus.

October 17, 1929.

3. The Measurement of Sound-Absorption by Oscillograph Records, V. L. CHRISLER, *Bureau of Standards* (20 minutes).

A sound decay curve in a reverberation room was photographed by means of an oscillograph. Computing the rate of decay from this film the total absorption in the room can be computed. The average results obtained by this method may be in close agreement with those obtained by the ear using the usual reverberation method, but the method as at present developed requires a prohibitive amount of time and labor.

4. An Audiometric Method for Measuring Sound-Insulation, WALLACE WATERFALL, *The Celotex Company* (20 minutes).

The audiometric method of measuring the sensation level of sounds can be readily adapted to sound insulation measurements. The equipment consists of an electrical oscillator and amplifier which, by an automatic switch, is connected alternately to a loud speaker and to an attenuator and telephone receiver. The loud speaker is placed on one side of the partition to be measured and, by varying the attenuator a loudness match is made between the sound from the loud speaker and from the telephone on each side of the partition. The attenuator is graduated in decibels so the difference in the readings gives the reduction factor of the partition. Tones of 256, 512, 1024 and 2048 cycles are used. For field work, the audiometric method has the advantage that noises usually encountered do not seriously interfere with measurements.

As a check on the audiometric method, eight adjoining rooms, each an eight foot cube, have been constructed. Walls of different types separate adjoining rooms. Measurements on these partitions by the audiometric and an instrumental method show good agreement. They also agree well with measurements by the reverberation method on an identical partition tested by Paul E. Sabine at Riverbank Laboratories.

November 6, 1929.

5. The Effects of Noises—An Abstract of Experimental Findings from Various Sources, DONALD A. LAIRD, *Colgate University* (20 minutes).

A survey of experimentation dealing with effects of various acoustic stimuli on motor functions, simple cerebral functions, complex cerebral functions, respiratory functions, cardiovascular functions, feeling-tone and emotional functions, and on the organism at work. Some thirty experiments cited show that acoustic stimuli have a wide variety of effects on mental and physiological functions and help explain general reactions of people to noise. Way for future development of more relevant studies shown.

6. The "Strike Notes" from Bells, ARTHUR TABER JONES, *Smith College* (20 minutes).

The "strike note" of a bell does not appear to be produced by vibration of the bell in any of its normal modes. The strike note is not far from an octave below the fifth partial tone of the bell, and the suggestion has been made that this fifth partial may come out with especial promptness and strength, and the attention of the ear be afterward quickly caught by lower tones, so that the tone from the fifth partial is judged to be an octave lower than it really is.

Several tests of this hypothesis have been attempted. (1) A number of musicians have given their judgment as to the pitches of the strike notes on three bells. The average is almost exactly an octave below the fifth partial. (2) The sounds from six bells have been picked up from a distance of about 300 meters. When the bells were struck so gently as to be almost inaudible at this distance, beats from suitably tuned forks showed that the fifth partial is at least as prominent as any other. (3) On the same six bells the first order difference tone from the fifth and seventh partials is close to the strike note. (4) The time during which the clapper is in contact with the bell has been found rather roughly for one bell. This time is not far from half the period of the fifth partial.

The results of other tests are not yet sufficiently definite to be of significance. The work is to be continued next summer.

SATURDAY, DECEMBER 14

9:30 A.M.

1. **Symposium on Theories of Hearing.**

   (a) GEORGE E. SHAMBAUGH, M.D., Chicago, Illinois. Anatomical Considerations. (25 minutes).

   (b) L. T. TROLAND, *Technicolor Motion Picture Corporation*, Physiological Considerations. (25 minutes).

   (c) HARVEY FLETCHER, *Bell Telephone Laboratories*, Physical Considerations. (25 minutes).

   (d) G. W. STEWART, *University of Iowa*. Binaural Hearing. (25 minutes).

   Discussion—

   A. G. POHLMAN, M.D., *St. Louis University*.
   F. W. KRANZ, *Riverbank Laboratories*.

2. **The Influence of Phase on Tone Quality,** W. C. BEASLEY, *Ohio State University* (20 minutes).

A special generator was built to permit alteration of phase between two fundamentals to a known degree, with an output of separate currents permitting inspection of the frequency ratios 15:16, 9:10, 2:3, 5:8, 3:5, 9:16, 5:9, 1:2, 1:3, 5:16. A useful formula is developed to predict the amount of phase retardation in the component of higher frequency necessary to effect a maximum change in compound wave form due to phase retardation, maintaining amplitude and frequency in the components constant. A differential gear driving mechanism is constructed to effect the critical amount of phase change during the same period of time for all frequency ratios. The results are contrary to those found by any former investigations. The conclusion is established that with change of phase, the relative loudness of the partial tones varies in opposite directions at the same time, the amount of change in relative loudness being roughly inversely proportional to the magnitude of the ratio terms of the fundamentals when these ratio terms are reduced to unity difference; further, the amount of change in relative loudness is a function of auditory sensitivity at various frequency levels. Tests were made over the frequency band 450 to 2400 cycles, at various sensation levels.

12:30 P.M. to 2:00 P.M.

Luncheon

SATURDAY, DECEMBER 14

2:00 P.M.

1. **The Use of Sound in Navigation,** HORATIO W. LAMSON, *General Radio Company* (20 minutes).

Modern methods for employing under-water sound as an aid to navigation are based upon the binaural sense.

The earliest devices, exemplified by the "C" tube, etc., mechanically placed the ears of the observer on the extremities of a beam to be rotated in a horizontal plane beneath the sea. Such cumbersome equipment was improved by employing fixed under-water receivers in conjunction with an adjustable compensator for obtaining a binaural balance.

Electrical lag lines have replaced acoustical tubes for obtaining the necessary time lag to affect compensation. The effectiveness of such a two-spot system is hampered whenever more than one sound source is present. This difficulty was subsequently overcome by obtaining a directional sensitivity by means of a multi-spot system which compensates simultaneously for a binaural and maximum focus.

Observations upon the compensators may likewise be made upon sounds traveling in a vertical plane. This enables continuous depth soundings to be taken with the vessel under way at full speed, a valuable feature.

Apparatus has also been perfected for acoustic depth soundings by timing the echo of sound reflected from the sea bottom. The use of super-audible sound waves, which can be projected in a slightly diverging beam, offer great possibilities for the location of icebergs, etc.

2. A New Acoustic Analyzer—Determination of the Sound Spectra Produced by Aircraft in Flight, L. P. DELSASSO, *University of California at Los Angeles* (20 minutes).

An Electro-Mechanical Frequency Analyzer, previously reported (Physical Review, August 1, 1929) has been adapted to the determination of the sound spectra of aircraft in flight. The complex source of sound is converted into potential fluctuations by a microphone and high quality amplifier. The output of the amplifier is applied to a modified quadrant electrometer, the needle of which forms part of a sharply resonant mechanical circuit. The natural period of the mechanical circuit is made continuously variable by a tri-filar suspension and can be determined from a single micrometer reading. An analysis of a complex source is made by observing the amplitude of vibration and natural period of the mechanical circuit for each component. By a previous calibration of the instrument these observations are sufficient to determine the amplitude and frequency components of the complex sound. The instrument has been used to measure the sound spectra of typical passenger and military planes in flight.

3. Measurement of the Constants of a Single Reed Wind Vibrator, R. L. WEGEL and R. R. RIEZ, *Bell Telephone Laboratories* (20 minutes).

Single reed wind instruments are represented by certain organ pipes, by the harmonica, clarinet, saxophone, duck call and the artificial larynx. In all of these the sound is produced by the periodic modulation of a stream of air which, when passing through a "glottis" or narrow passage between the reed and a juxtaposed stationary surface or "bed," energizes the device. The forces concerned are the attraction of the reed to its bed due to the constriction of the air stream at the glottis according to the principle of a venturi meter, the pressure of the air on the reed and on the air stream in the glottis. Vibration results when the air flow is such that the proper relative magnitudes and phases of these forces occur. These forces have been measured in a model artificial larynx and the calculated vibration conditions checked against measurements of pitch and amplitudes.

October 17, 1929.

4. Absolute Amplitudes and Spectra of Certain Musical Instruments, L. J. SIVIAN, H. K. DUNN, and S. D. WHITE, *Bell Telephone Laboratories* (20 minutes).

Systems for the reproduction of music are limited in the frequency range and in the amplitude of the sounds which they can reproduce without distortion. In order to determine what requirements such a system should meet for faithful reproduction of common forms of music, those instruments which were believed to contribute largely to low or high frequencies, or to have large amplitudes at any frequency, have been investigated. Those examined were: bass drums of three sizes, snare drum, cymbals, triangle, bass viol, bass saxophone, bass tuba, trombone, trumpet, French horn, clarinet, flute, piccolo, piano, and organ. Several measurements have also been made upon complete orchestras.

The measurements included average and peak sound pressures corresponding to the whole spectrum, and also the same quantities in thirteen bands of frequencies which together covered the complete spectrum. Several hundred peak measurements were taken in each band, each measurement representing the highest peak in one-eighth second of music. They were taken in such a way that the frequency of occurrence of the peaks in ten amplitude zones, each zone covering six decibels, was obtained. The musician in each case played a representative selection for his instrument, lasting two or more minutes.

With the complete orchestra, the highest peaks were found to lie in the region between 2,000 and 4,000 c.p.s. The greatest peaks below 64 c.p.s. were obtained when the bass drum was prominent, while the greatest peaks above 8,000 c.p.s. were very plainly due to the cymbals.

October 17, 1929.

5. Music as a Humanizing Agent, HENRY C. LOMB, *Music Industries Chamber of Commerce* (20 minutes).

Among the many possible applications of acoustics, the practical application of music to the humanizing or betterment of all elements of our populations stands in the front rank.

Such application presupposes an influence of music on human behavior. Of this influence we find daily corroboration in the playing of martial and sacred music and in the melodies and harmonies of song and symphony.

The cause of this effect has until recently baffled scientific investigation. But the tremendous potency of music is doubtless largely due to its appeal to all classes of humanity. As the sole universal language of mankind, far transcending speech in range and power, it should be possible to employ it as a great humanizing force for the benefit of mankind.

The effectiveness of music is not limited here by its character of a language of feelings, since all mental and physical activities are determined greatly, even though unconsciously, by our emotions.

Research, particularly in the psychology of music, to bring about larger actual and personal participation in musical activities, not merely passive contemplation thereof, is here indicated.

6. The Approximate Networks of Acoustic Filters, W. P. MASON, *Bell Telephone Laboratories* (20 minutes).

The original theory of acoustic filters, given by Stewart, is based on the representation of such filters by means of lumped constants in the form of a T-network. More recently, the writer has presented a theory of acoustic filters, showing that they are equivalent to a combination of electric lines. Stewart and Lindsay have pointed out that the first theory is a first

approximation of the second theory—obtained by taking the first terms of the trigonometrical expansion occurring in the latter theory.

This approximation holds well for low pass filters, at low frequencies, but does not represent very adequately the band pass filter. Accordingly a second approximation is developed in this paper. Methods are given for determining the transmission characteristics of any types of acoustic filters. These methods have been applied to analyzing all of the single band pass filters possible with one main branch and any side branch combination. There are five of these filters, and design formulae for them have been worked out. It is possible, from these formulae, to determine the dimensions of a filter necessary to have a given attenuation and impedance characteristic.

November 11, 1929.

7. Sound Concentrator for Microphones, HARRY F. OLSON and IRVING WOLFF, *Radio Corporation of America* (20 minutes).

A sound concentrator is described having the shape of a parabolic horn, with a microphone at the focus and an associated acoustic network for equalizing the response. Amplification is obtained at low frequencies due to action as commonly attributed to horns and at high frequencies due to the reflections of sound from the walls to the focus. The theory for the parabolic horn attached to a microphone is developed and compared with the experiment.

In order to determine the frequency at which the reflection from the walls becomes of importance, a comparison of the parabolic horn with a double conical horn of approximately the same size is made. Experimental results on these two horns show that reflection from the walls begins at 1500 cycles and increases in importance with frequency. The directional pick-up properties of this parabolic horn have been investigated and determined as a function of the frequency of the impinging sound. The characteristic peaks of the short parabolic horn are reduced to an optimum value by means of Helmholtz resonators loosely coupled to the horn at the receiver end.

SUPPLEMENTARY PROGRAM

An Error in the Derivation of the Reverberation Formula, HALLET H. GERMOND, *C. F. Burgess Laboratories*,

Sabine's formula $T=55.3V/\text{avs}$ yields finite values for the period of reverberation even when the coefficient of absorption is unity. This formula is a first approximation to that developed by Norris* $T=-24V/\text{vs}\ \text{Log}_{10}\ (1-a)$; the latter yields $T=0$ for $a=1$. An error in the interpretation of the fundamental differential equation in the derivation of Sabine's formula is demonstrated.

Another mode of attack permits a more correct analysis of a room under certain specified distributions of absorbing material.

November 14, 1929.

Optimum Reverberation Time for Auditoriums, W. A. MACNAIR, *Bell Telephone Laboratories*.

Up to the present, the acoustics of auditoriums has rested on a purely empirical basis. A more satisfactory situation would be reached if we knew a physical quantity which had some definite value for all rooms, regardless of size, which are considered acoustically good. Such a condition is that

$$\int_{t_0}^{t_1} L_t\, dt = -32.6$$

where t_0 is the time a sustained source of sound, $\overline{E}$, is cut off; t, the time the sound becomes inaudible; and L_t, the loudness of the sound at any instant t. For a test tone of 1000 cycles we take $\overline{E}$ to be 35.3×10^{10} cubic feet of sound of threshold density per second, which is a fair average of sources met with in practice. All rooms which have been considered acoustically good satisfy this condition. This implies that the ear is a ballistic instrument during the period of hang over of a sound, and suggests that rooms should be adjusted so that the loudness of all pure tones shall decay at the same rate, independent of frequency. The absorption material used must have a definite frequency characteristic to obtain this result. This characteristic turns out to be very near to that which an audience of people exhibits. An audience is recognized as being nearly ideal absorption material.

* Presented during discussion at New York meeting of the Society May 10, 1929 by R. F. Norris.

MEASUREMENT OF SOUND TRANSMISSION*

By V. L. Chrisler
Bureau of Standards

Some of the most difficult and yet most important problems in architectural acoustics are associated with the sound-proofing of walls and floors. Difficulties are encountered both in designing an approximately sound-proof structure, and in making measurements of its sound transmission.

Let us consider first the problem of making the measurements. For this purpose the first requisite is a suitable source of sound. The Bureau of Standards maintains that the most satisfactory source of sound is one giving a pure note free from overtones. This seems as axiomatic as the use of pure reagents in chemical analysis. With the same panel, the transmission may be quite different for notes of varying pitch. If therefore a complex tone is used the transmission will vary with the composition of the tone; and unless this composition is specified the results will be of little use.

It is by no means easy to specify and reproduce a complex tone, but it is practicable to produce tones of a high degree of purity. Such tones are obtained at the Bureau by means of a beat frequency oscillator and a 3-stage resistance-coupled amplifier. The output from the amplifier is fed into a high-grade dynamic loud-speaker. The purity of the tone is checked by oscillograph observations, and it is found that as long as the tubes are not overloaded very good results are obtained.

Having obtained a suitable source of sound, the next problem is to find a satisfactory method of measuring the fractional part of the sound which passes through the panel. Here it is desirable to have some form of instrument for making such measurements.

There are two types of such instruments available, the mechanical, such as the Rayleigh disc, and the electrical. Instruments of the latter class usually consist of a telephone or microphone as a pick-up and a suitable indicating instrument aided by an amplifier.

The Rayleigh disc is not a very sensitive device at its best, while the electrical type of instrument can be made highly sensitive. Moreover the latter type is portable, while the Rayleigh disc cannot be readily moved around.

* Publication Approved by the Director of the Bureau of Standards of the U. S. Department of Commerce.

However, the electrical type of instrument has its troubles. If an ordinary telephone receiver is used it is found to be much more sensitive to tones of a frequency around 500 cycles and over than to lower notes. In fact, the ordinary telephone may be quite insensitive from 150 cycles down. This means that slight overtones may be greatly overemphasized.

The condenser microphone has a much wider range of nearly uniform response, and is therefore a preferable instrument for a pick-up.

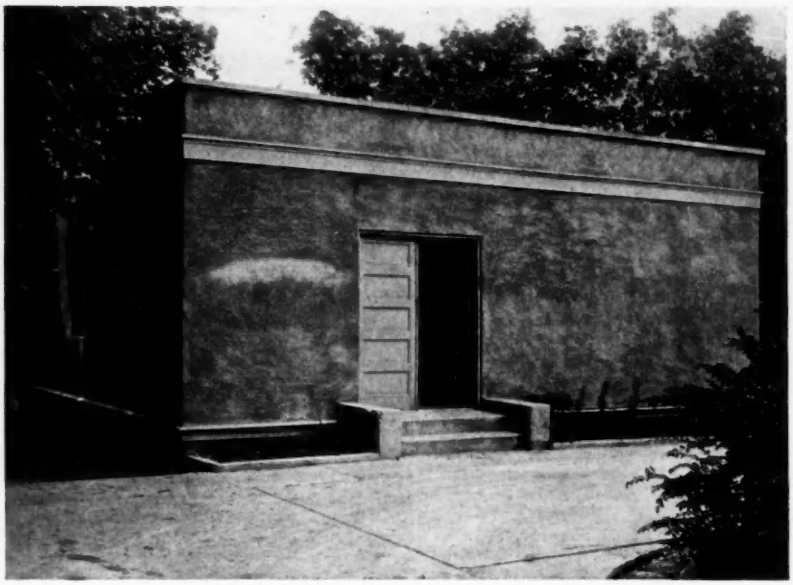


FIG. 1. *Sound Chamber, front view.*

The special laboratory used for sound measurements at the Bureau of Standards is shown in Fig. 1. A vertical cross-section is shown in Fig. 2. The transmitting room (containing the loud-speaker) is built with separate foundations, walls and ceiling. Floor panels to be tested are laid in the upper opening and wall panels are placed vertically in an opening in one of the walls.

With no panel in position there is some reduction in the intensity of sound in passing through the empty opening. This reduction is determined and applied as a correction to the final measurement.

The panel is then placed in position in the opening and carefully sealed around the edges. The average sound intensity is now redetermined on both sides of the panel, and from these measurements the fractional part of the sound transmitted is computed.

It should be noted that the absorption characteristics of the receiving room may affect the results. To go to extremes, suppose there is no absorption of sound by the walls of the receiving room. In this case the level of intensity will build up until as much sound passes through the panel in one direction as in the other, and there will be no reduction in intensity produced by the panel. On the other hand, if the walls of the receiving room were perfect absorbers of sound the same panel might show a large reduction factor. This is one of the reasons why trans-

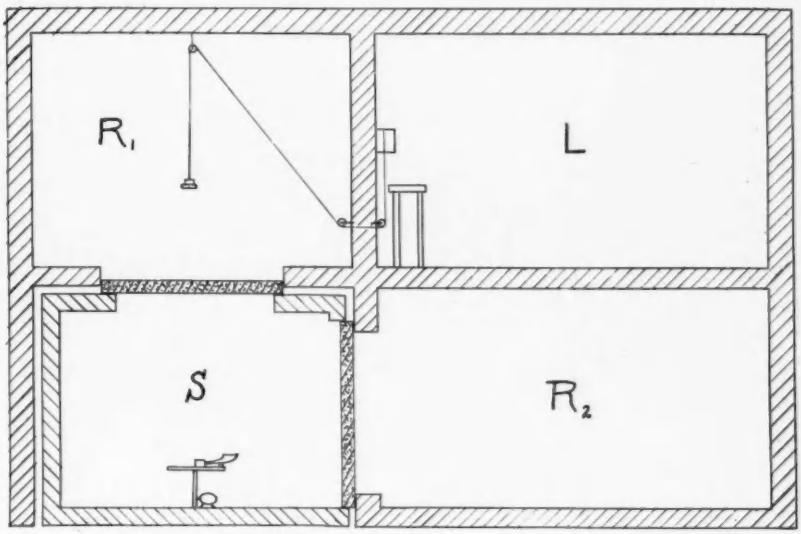


FIG. 2. *Sound Chamber, Verticle Section.*

mission results obtained by different laboratories do not agree. And even if some definite absorption for the receiving room were agreed upon, the results given by panels when actually placed in buildings might differ to some extent from their laboratory values. Experimental investigation of this point has been started.

Fig. 3 shows some of the measurements made on materials of a more or less homogeneous structure. The lightest material tested was wrapping paper, and among the heaviest were an 8 inch brick wall and a combination floor slab. It will be seen that over this great range of weight, about 1 to 10000, the reduction factor is approximately proportional to the logarithm of the weight per unit area.

Fig. 4 shows the results obtained by Dr. Sabine and Mr. Heimburger compared with those of the Bureau of Standards. These curves will

show the straight-line relation just pointed out, and also indicate that the results are affected by the method used or by the character of the

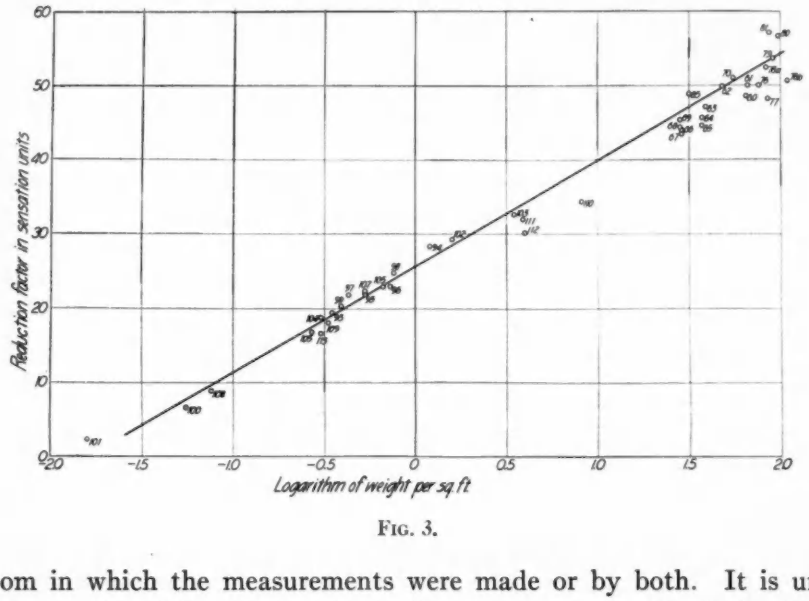


FIG. 3.

room in which the measurements were made or by both. It is unfortunate that it is at present impossible to compare results obtained by different observers.

In addition to transmission measurements made with pure tones, talking tests were also made by having one observer attempt to speak

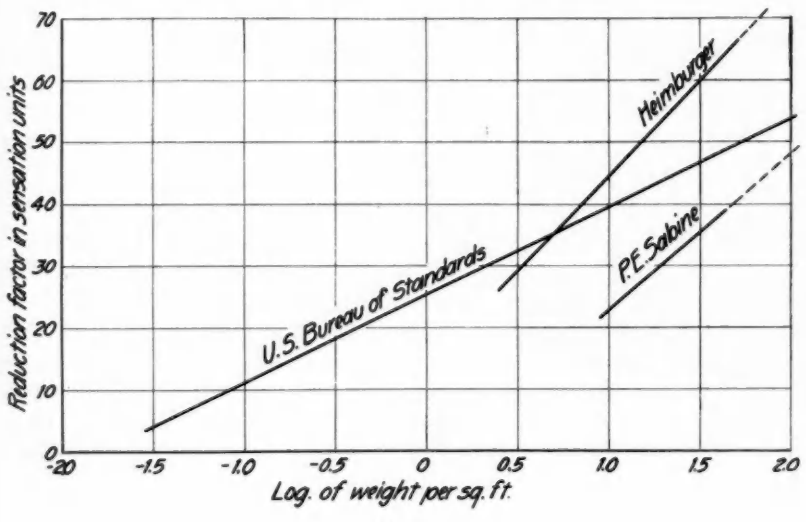


FIG. 4.

to a second one on the opposite side of the panel. It was found that complete suppression (under laboratory conditions) of the sound of a conversation carried on in an ordinary tone of voice would require a panel with a reduction factor of not less than 60 decibels. Referring again to Fig. No. 3, we find that the heaviest masonry wall tested does not quite fulfill this requirement. It is therefore evident that weight alone can never be satisfactorily employed in the attempt to sound-proof partitions in modern structures.

In seeking a solution of this problem a large number of experiments have been made. The results, stated briefly, are as follows: The first consideration should be the nature of the framework, and for this masonry is the best type as it has considerable inertia. The next consideration is the inside finish of the room. The ideal plan would be to have walls, ceiling and floor form practically an inner room, separated by a space of an inch or so from the masonry structure. This is rather impracticable. If some kind of support could be devised to hold these surfaces and yet not permit vibrations to be transmitted to the frame the problem would be solved.

The best wall structures tested at the Bureau had a 4″ tile core with wood furring strips to which were attached the plaster base and the plaster. The furring strips were wired to the masonry core. It is believed that this rather imperfect connection contributed largely to the excellence of the result. In such a structure one surface was free to vibrate to some extent without setting the core or the other surface in motion.

Two systems of sound-proofing now in the market make use of this general idea. In one system the surface is supported by chairs lined with hair felt; in the other steel springs are employed. No comparative tests of these two systems with ordinary structures have been made by the Bureau.

In many ways floor structures present a particularly difficult problem, as we have to deal in this case not only with air-borne noises but with impacts produced by walking or by moving furniture. This kind of noise is much more difficult to deal with than air-borne noises. For experimenting with impact noises the machine shown in Fig. 5 was used. The five weights shown are lifted by cams and allowed to drop one by one at intervals of about one-fifth of a second.

When the machine operated directly on the face of a 4″ concrete slab, the transmission was almost perfect. With a floating floor on top of the slab and a suspended ceiling below, the reduction in intensity was

about 51 decibels. It is believed that this would be sufficient to prevent the transmission of the sound of footsteps under ordinary conditions. The best way to eliminate such noises, if possible, is at their source, by the use of sufficiently thick rugs. Unfortunately this is not always possible.

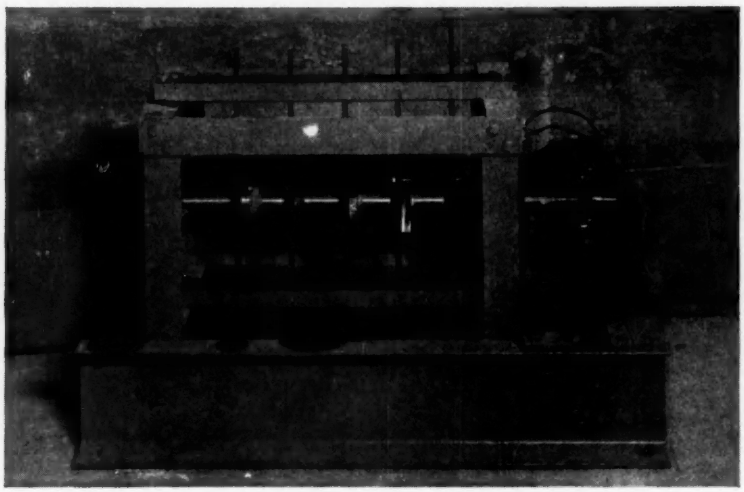


FIG. 5. *Machine for Producing impact Sounds.*

Improvement of wood structures seems rather a hopeless task. One method that has been used for such walls is to stagger the studs, but our tests showed an improvement of only 4 decibels over the ordinary type of construction. Other attempts have been equally discouraging. Our best results have been obtained with a plaster which has very little mechanical strength. However, this type of construction is not practical, because the life of such a plaster would be very short.

From the results given it is evident that sound-proofing is not so much a matter of materials as it is a matter of structure. If any progress is to be made along the lines of sound insulation it is to be made by studying structures or combinations of materials rather than the materials themselves.

These remarks may perhaps have given a rather gloomy picture of the situation. The problem is indeed difficult and much work remains to be done. But sound insulation is not unattainable. It can be realized if sufficient care is taken and if cost is not too great a consideration. The real problem at present is how to obtain a reasonable degree of sound insulation without appreciably increasing the cost of construction.

TRANSMISSION OF SOUND BY WALLS

By Paul E. Sabine
Riverbank Laboratories, Geneva, Ill.

The investigation of the problem of transmission of sound between adjacent rooms by way of intervening partition was begun at the Riverbank Laboratories in 1919. The study of the problem has been conducted with a three fold purpose in mind; (1) to determine the various physical properties that affect the passage of sound by wall construction and their relative importance (2) to make quantitative determinations under constant conditions of the degree of acoustic insulation afforded by normal types of partition, and (3) to discover if possible practicable means of increasing sound insulation in buildings. Reports on various portions of the work have been made from time to time.[1]

It is the purpose of the present paper to give in summarized form the outstanding facts which have been brought out, presenting the experimental results in detail only where these serve to illustrate considerations which are significant in the general problem of sound insulation.

Method and Conditions of Tests

The method used throughout may be called the Reverberation Method, and is described in the earlier papers. It is based upon the theory of reverberation in rooms developed theoretically and experimentally by Professor Wallace C. Sabine. If an average sound intensity, (sound energy per unit volume) I_1 be set up inside a closed space, then after the source ceases the decrease of intensity due to sound absorption at the boundaries of the room is given by the equation

$$\log_e \frac{I_1}{I} = At$$

in which I is the intensity t seconds after the intensity was I_1, and A is a constant involving the volume of the room, its total absorbing

[1] The American Architect, July 28, 1920, Transmission of Sound by Doors and Windows.
The American Architect, Sept. 28, 1921, Transmission of Sound through Flexible Materials.
The American Architect, July 4, 1923, Transmission of Sound by Masonry Materials.
The American Architect, August 5, 1926, Transmission of Sound by Wood Stud Partitions.
The Armour Engineer, May 1926, Transmission of Sound by Double Partitions.

power measured in square units of perfectly absorbing surface and the velocity of sound. If i is the threshold intensity of sound, that is the least intensity that is audible to a given observer, and t_1 the time required for sound of initial intensity I_1 to decrease to this threshold, we have

$$\log_e \frac{I_1}{i} = At_1.$$

For the Riverbank Sound Chamber, this equation reduces to the form

$$\log_{10} \frac{I_1}{i} = .130\,at_1, \tag{1}$$

in which a is the total absorbing power of the Chamber under the test conditions. The various methods used for the calibrations of this room, that is the determination of a for each of the various test tones used was described in detail in the Journal of the Franklin Institute for March 1929.[2]

The partition undergoing test is built into an opening between the Sound Chamber and a structurally separate, adjacent smaller room, the Test Chamber. To eliminate the effect of reverberation in the Test Chamber a small portion next the partition is walled off with a highly absorbent material. The observations consist in measuring the time t_1 in the Sound Chamber and the time t_2 for the same sound heard in the Test Chamber through the partition.

The sound insulating merits of a partition for a tone of given pitch is expressed numerically by the so-called Reduction Factor denoted by the letter k. If I be an intensity in the Sound Chamber which is barely audible in the Test Chamber on the other side of a partition having a reduction factor k, then

$$I = ki.$$

Thus a wall with a Reduction Factor of 1000 would render inaudible any sound whose intensity is 1000, or less than 1000, times the threshold intensity for the particular observer.

Suppose now that a sound of initial intensity I_1 can be heard for t_1 seconds in the Sound Chamber and for t_2 seconds on the further side of a partition, the Reduction Factor of which is k

Then

[2] The later calibrations of the Sound Chamber gave values of a somewhat higher than the earlier calibration. All values of sound reduction given in this paper are based on this later calibration and are therefore slightly greater than earlier published figures.

$$\log_{10} \frac{I_1}{ki} = .130\, at_2 \qquad (2)$$

Subtracting equation (2) from equation (1) we have

$$\log_{10} k = .130\, a(t_1 - t_2) \qquad (3)$$

Thus defined and measured *k is the least intensity measured in terms of the threshold intensity which can be heard* on the further side of the partition. It will be noted that its value as thus measured is independent of the absolute value of the observer's threshold intensity. Experiments show very good agreement between the results obtained by different observers, so that while the method is based upon aural observations, yet the rather wide variations known to exist between the sensitivities of hearing of even normal ears do not affect the measured values.

Effects of Pitch

It was early recognized that the value of the Reduction Factor for a given partition varies markedly with even slight variations in pitch. Speaking generally, it is greater for high than for low pitched sounds. In order, therefore, to arrive at conclusions that should be of general validity, the tests have covered the whole range of frequencies from 128 to 4096 vibrations per second. In view of the more frequent occurrence in practice of intense sounds in the lower range and of the greater variation with pitch for the lower tones, the standard tests have included 4 tones in each octave below 1024 vibs/sec. and two tones in each of the two octaves above this frequency, 17 tones in all. In order to express the results of the tests by a single numerical value, the standard practice of taking the average of the logarithm of the Reduction Factors for all the test tones has been adopted. These average values for the various partitions tested are taken as the measure of their relative sound insulating merits.

Doors and Windows

A door or window may be treated as a single structural unit, and the tests on these constructions show that the reduction factors are determined for the greater part by the weight of such units and also, but to a less degree, by their structural stiffness. Table I gives the averaged results for 16 different units. Unless otherwise indicated the units were

TABLE 1.

No.	Description	Ave. Reduction (dbs.) 128 to 4096	128 to 1024
1.	Steel, $\frac{1}{4}''$ thick	30.5	30.0
2.	Refrigerator Door $5\frac{1}{2}''$ thick Yellow Pine filled with cork	25.8	24.6
3.	Solid Oak, $1\frac{3}{4}''$ thick	22.0	20.9
4.	Fabricated, hollow, flush or sanitary door $1\frac{3}{4}''$ thick	23.6	22.7
5.	No. 4 hung in usual manner	21.2	20.8
6.	No. 4 with layers of $\frac{1}{2}''$ Celotex in hollow space	23.6	22.5
7.	No. 4 with $1''$ Balsam wool in hollow space	23.4	22.0
8.	Light, 4-panel, birch veneer	19.2	18.6
9.	Two doors, panelled maple veneer, hung with $2''$ separation in single casement	26.4	25.0
10.	Window, one pane $79''\times30''$, $\frac{1}{4}''$ plate glass	23.0	22.8
11.	Window, 4 panes each $15''\times39''$, $\frac{1}{4}''$ plate glass	25.7	25.3
12.	Window, 2 panes each $31''\times39''$, $\frac{3}{16}''$ plate glass	20.1	19.7
13.	Same as 12, but double glazed with glass set in putty on both sides of sash, $1''$ separation	23.4	23.1
14.	Same as No. 13 but with glass set in felt	25.4	24.0
15.	Small $\frac{3}{16}''$ leaded panes	25.0	23.9
16.	12 panes $10''\times19''$, $\frac{1}{8}''$ window glass	21.7	21.2

sealed in tightly in each case so that the values are somewhat higher than would be the case in actual practice, when the sound insulation is decreased by the passage of sound through the cracks at the threshold and sides. Comparison of Nos. 4 and 5 shows the order of magnitude of the effect of the usual clearance necessary in hanging a door. Comparison of No. 4 with No. 6 and No. 7 indicates the ineffectiveness of filling a cellular space within a structure with a light sound absorbent material. Inspection of the values for the windows suggests that the cross-bracing of the sash effects a slight increase in insulation over that afforded by larger unbraced areas of glass. On the whole, it may be said that normal door and window construction can not be expected to give a reduction of more than 25 db.

FLEXIBLE, POROUS, MATERIALS

Transmission of acoustic energy by partitions of wood, glass, steel or masonry, which are impervious, must take place by means of the minute vibrations of these structures set up by the alternating pressure of the incident sound. Sound, *considered as undulatory motion of the air particles cannot pass through such barriers.*

For this reason, the gross mechanical properties of mass, stiffness

and internal friction or damping of these constructions determine the reduction of sound intensity which they afford. On the other hand, porous materials allow the alternate condensations and rarefactions to pass through, so that such materials may be said truly to transmit sound as sound. One is thus led to expect that the sound transmitting properties of such materials will differ from those of solid impervious materials. And this is the case. Summarized these differences are as follows: (a) In porous materials, the Reduction Factor varies in a continuous manner with the pitch. (b) At a fixed pitch the increase in the logarithm of the Reduction Factor for a porous material is directly proportional to increase in the thickness. This latter fact is expressed by the equation

$$R = 10 \log k = 10(r + qx)$$

in which x is the thickness, a is a constant which is related to the absorption coefficient of the material, and b is a second quantity which is constant for a given pitch and a given material. The form of this equation indicates that the loss of energy in transmission is a true absorptive process. From the equation and the experimentally determined values of r and q one can compute the reduction in decibels produced by any thickness of a given porous material. It should be said that the foregoing, as well as the figures given in Table 2, applies

TABLE 2.

FLEXIBLE MATERIALS

No.	Reduction dbs.
17. Hair felt 1″ thick	6.3
18. " " 2″ "	9.3
19. " " 3″ "	11.8
20. " " 4″ "	14.7
21. 3 layers 1″ felt alternated with 4 layers building paper	27.9
22. 4 layers of ½″ Cabot quilt	19.4
23. 4 layers of ½″ Flaxlinum	26.1

only to the case in which the porous material alone acts as an intervening partition. The introduction of impervious layers such as building paper or steel, entirely alters the character of the phenomena. Moreover as will subsequently appear, there is little or no significance to be attached to the results of tests conducted on porous materials alone, when applied to cases in which they are incorporated in other-

wise rigid construction. In such cases sound reduction is dependent upon the mechanical properties of the construction as a whole rather than upon the insulating properties of its components. Table 2 gives the Reductions shown by a few of the combinations tested.

CONTINUOUS MASONRY

By "continuous masonry" we shall mean single walls, as contrasted with double walls, of solid or hollow clay, or gypsum tile, solid plaster on metal lath and channels, and brick. These include most of the usual

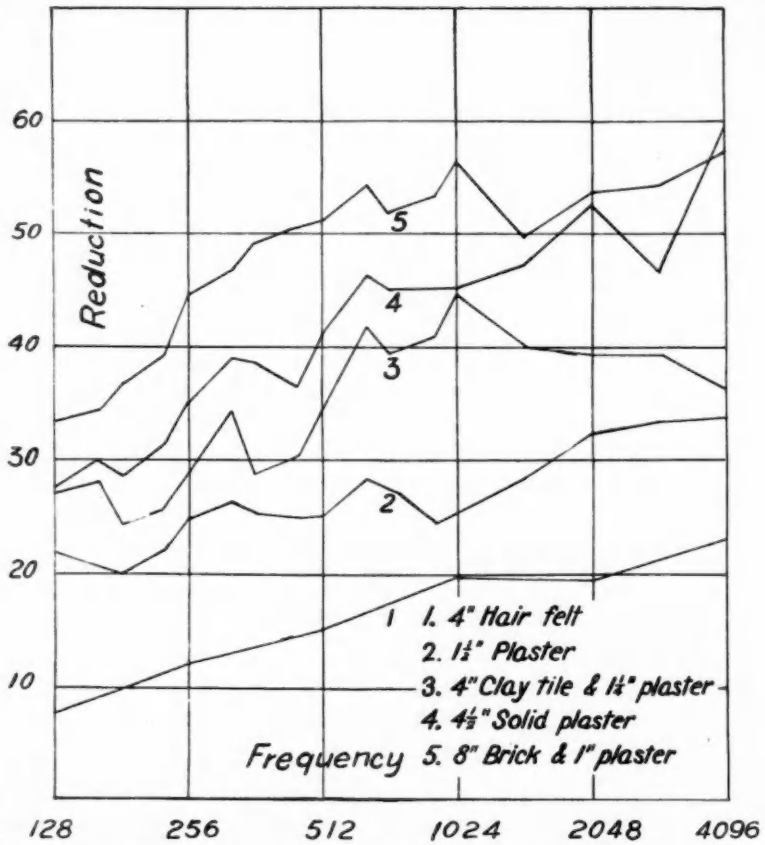


FIG. 1. *Reduction by continuous walls of sound of different frequencies.*

types of normal masonry partitions. In Fig. 1, Reduction is plotted against the pitch of the sound for four different masonry constructions, and also for purposes of quantitative comparison a corresponding curve for 4 inches of standard hair-felt. In Table 3, are given the average

TABLE 3.

No.	Construction	Wt. per Sq. Ft.	Ave. Reduction 128–4096	Ave. Reduction 128–1024
24.	2″ Gypsum Tile, unplastered	10.4	22.8	20.2
25.	3″ Hollow Gypsum Tile, unplastered	11.1	24.0	22.3
26.	1½″ Plaster on metal lath	13.9	26.1	24.4
27.	3″ Solid Gypsum Tile, unplastered	14.2	27.6	23.7
28.	2″ Solid Gypsum Tile and ½″ plaster	15.0	27.2	24.8
29.	4″ Clay Tile, unplastered	17.0	29.2	28.8
30.	2″ Solid Gypsum Tile, 1″ plaster	19.6	30.1	26.6
31.	2″ Solid Gypsum Tile, 1¼″ plaster	21.4	31.0	28.9
32.	4″ Clay Tile, ½″ plaster	22.0	32.0	30.8
33.	2½″ Plaster on metal lath	23.2	33.3	30.5
34.	3″ Solid Gypsum Tile, 1¼″ plaster	25.4	34.3	32.6
35.	4″ Clay Tile, 1″ plaster	27.0	35.0	33.6
36.	4″ Clay Tile, 1¼″ plaster	28.6	35.6	34.1
37.	3½″ Plaster on metal lath	32.5	36.1	34.3
38.	4½″ Plaster on metal lath	41.8	39.8	37.9
39.	8″ Brick Wall, 1″ plaster	88.0	47.5	46.1

(Note. Gypsum plaster was used throughout this series.)

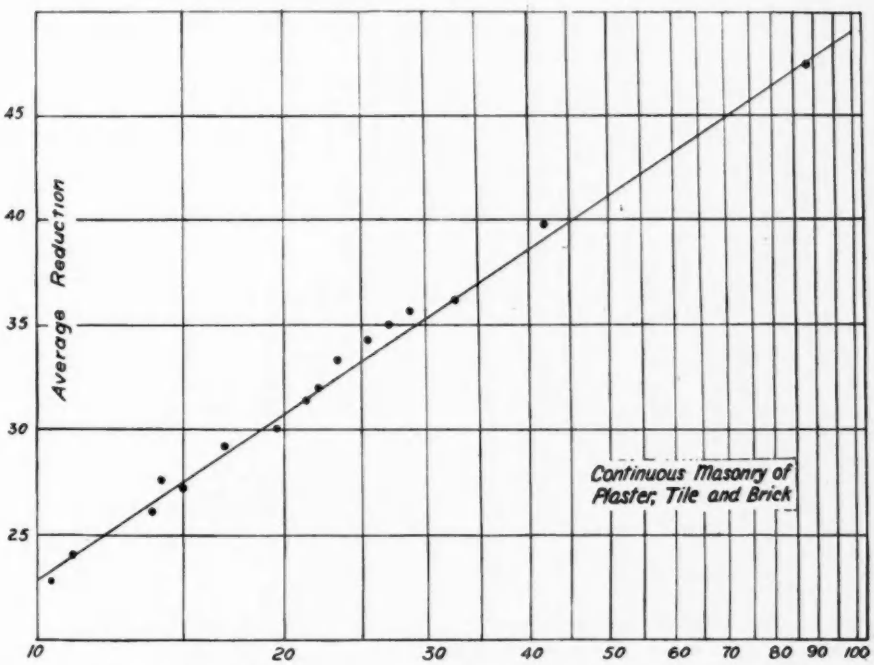


FIG. 2. *Sound reduction by continuous masonry plotted logarithmically against the weight per square foot.*

reductions for sixteen partitions of various masonry materials, together with the weight per square foot and thickness of each finished structure.

It will be noted that there is marked correspondence between the weights per square foot and the average Reduction. In Fig. 2, the average Reduction in decibels is plotted logarithmically against the weight per square foot, showing this correspondence in a very striking way. Writing down the equation of the straight line along which the expermental points lie, we have

$$R = 10(\log k) = (2.63 \log w - .34)10 \tag{4}$$

This may be thrown into the form

$$k = .456w^{2.63}. \tag{5}$$

It should be pointed out and definitely borne in mind that this is a purely empirical equation formulating the results of these experiments. As is indicated in Table 3, the average Reduction for each partition depends upon the range and distribution of test tones, so that the values of the two constants have little significance beyond the scope of these tests. The tests do however give a very fair degree of certainty to our expectations as to the degree of sound insulation possible with normal continuous masonry construction of a given weight, regardless of the particular masonry material used.

The results shown serve the further useful purpose of giving a basis of comparison for the merits of special types of construction designed for sound insulating purposes. In large buildings, allowable "dead load" fixes the upper limit of possible sound insulation using ordinary types of partitions. In practice, the sound insulating merits of any proposed special construction should be rated in terms of weight and thickness as well as of its reduction factor. For example, other things being equal a special construction weighing 20 pounds per square foot giving an average Reduction of 35.0 would have the practical advantage of less "dead load" over a 4″ clay tile partition weighing 27 pounds per square foot with the same reduction. Moreover, having determined the reduction for a particular type of construction, this relation enables us to state its sound insulating equivalent in inches of any particular type of solid masonry, brick, let us say. Thus a staggered wood stud, metal lath and gypsum plaster partition, weighing 20 pounds per square foot showed an average reduction of 39.1 db which is the same as that for a continuous masonry structure weighing 41 pounds per square foot, or, assuming a brick and mortar wall to weigh 120 pounds per cu. ft., the

staggered stud construction is equivalent to 4.1 inches of brick and mortar.

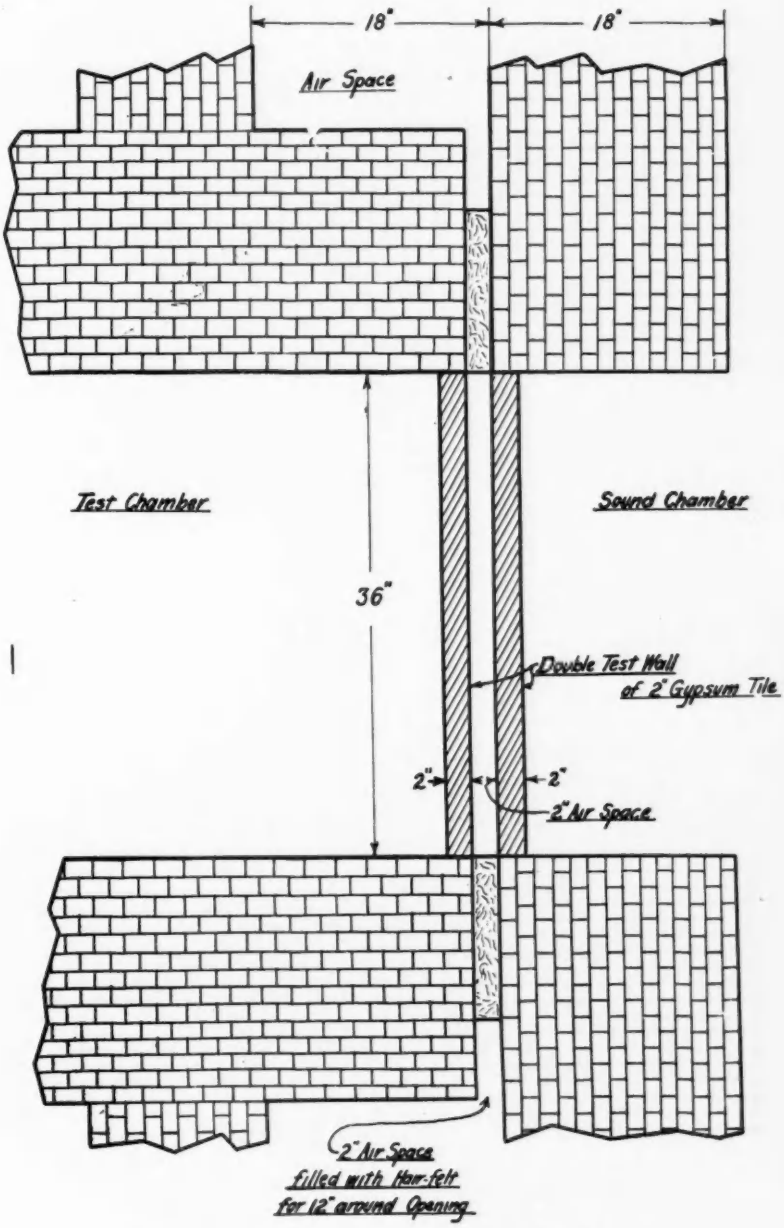


FIG. 3. *Detail showing complete structural separation of wall indicated in Table 4.*

DOUBLE WALLS, COMPLETELY SEPARATED

In practice, it is seldom possible to build two walls entirely separated. They will of necessity be tied together at the edges. The construction of the Sound Chamber and the Test Chambers is such, however, as to allow two walls to be run up with no structural connection whatsoever. (Fig. 3.) This arrangement makes it possible to study the ideal case of complete structural separation and also the effect of various degrees

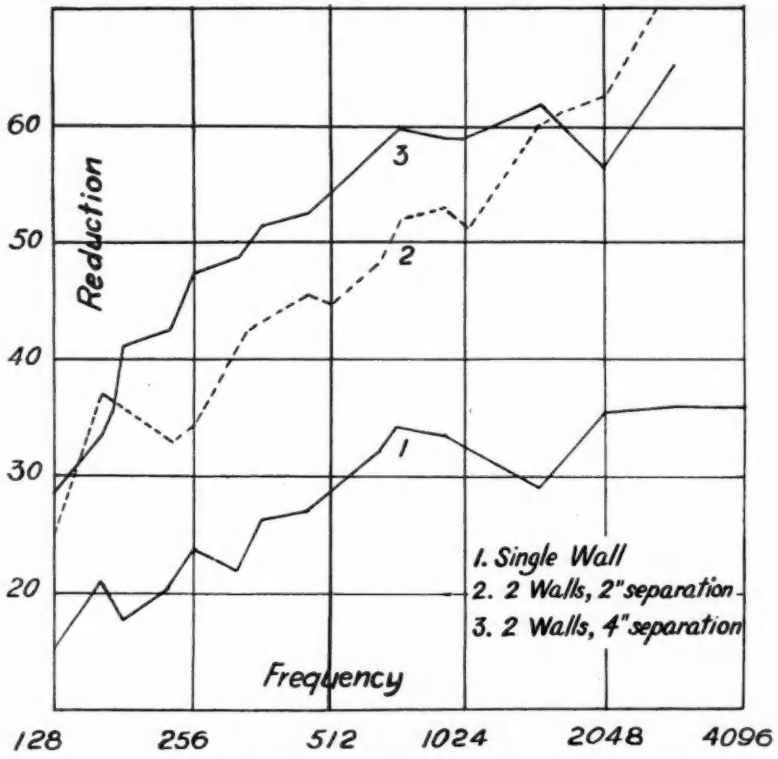


FIG. 4. *Graphs showing the effect of spacing in double wall constructions.*

of bridging or tying as well as that produced by various kinds of lagging fill between the walls. (Fig. 4) gives the detailed results of tests on a single wall of 2″ solid Gypsum Tile, and of two such walls completely separated, with intervening air spaces of 2″ and 4″ respectively. One notes that the increased separation increases the insulation, for tones up to 1600 vibs./sec. At higher frequencies, the 2″ separation is better. Another series of experiments shows that still further increase in the separation shifts the dip in the curve at 2048 vibs./sec. to a lower frequency. This fact finds its explanation in the phenomenon of resonance

of the enclosed air, so that there is obviously a limit to the increased insulation to be secured by increasing the separation.

(Fig. 5) shows the effect (a) of bridging the air gap with a wood strip running lengthwise in the air space and in contact with both walls, and (b) the effect of filling the inter-wall space with sawdust. One notes

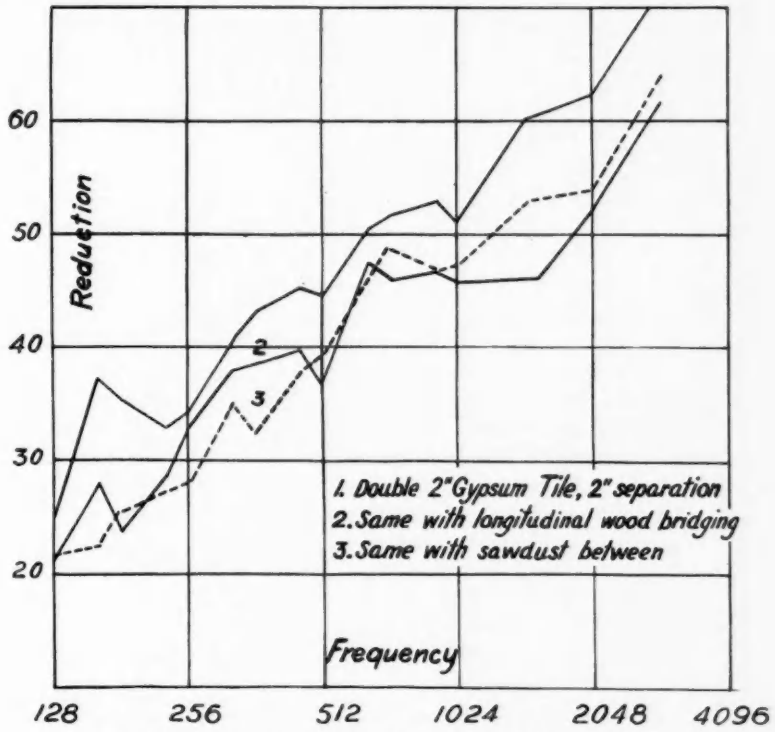


FIG. 5. *Graphs showing the effect of bridging a double wall construction.*

that the unbridged, unfilled space gives the greatest sound reduction, and further that any damping effect of the fill is more than offset by its bridging effect. Experiments with felt and granulated blast-furnace slag showed the same effect, so that one arrives at the conclusion, that, *if complete structural separation were possible*, an unfilled air space would be the most effective means of securing sound insulation by means of double partitions. As will appear in a later section, this conclusion does not include cases in which there is a considerable degree of structural tying between the two members of the double construction.

Table 4 gives the summarized results of the tests on double walls with complete structural isolation.

TABLE 4.

DOUBLE WALLS, STRUCTURALLY SEPARATED

No.	Specifications	Wt/area	Reduction	Equiv. Masonry
40.	Double 2″ Solid Gypsum Tile, unplastered, unbridged, 2″ separation	20.4	49.3	10.0″
41.	The same, bridged at middle	20.4	42.2	5.4″
42.	" " filled with sawdust	23.0	42.0	5.4″
43.	" " " " slag	30.9	42.7	5.7″
44.	" " " " felt	22.3	48.5	9.3″
45.	" " No. 40, but with 4″ separation	20.4	51.8	12.5″
46.	" " No. 45, but bridged top and bottom	20.4	46.9	8.1″
47.	Same as No. 45, but with inner faces lined with 1″ felt	22.3	57.3	

DOUBLE PARTITIONS, PARTIALLY CONNECTED.

Under this head is included types of double walls in which the two members are tied to about the same degree as would be necessary in ordinary building practice. In this connection, data showing the effect of the width of the air space may be shown. This series of tests was conducted with two single pane $\frac{1}{4}''$ plate glass window, $82'' \times 34''$ set in one of the Sound Chamber openings. Spacing frames of 1″ poplar to which $\frac{1}{2}''$ saddler's felt was cemented were used to separate the two windows.

The separation between the windows was increased by increasing the number of spacing frames. It is evident that the experiments did not show the effect of increased air space alone, since a part of the transfer of sound energy is by way of the connection at the edges. However,

TABLE 5.

DOUBLE WINDOWS, $\frac{1}{4}''$ PLATE GLASS

	Description	Reduction	Equiv. Masonry
48.	Sashes in contact	29.3	1.8″
49.	$1\frac{1}{2}''$ separation	34.0	2.6″
50.	$4\frac{1}{2}''$ "	35.3	3.0″
51.	$7\frac{1}{2}''$ "	38.9	4.1″
52.	$9\frac{1}{2}''$ "	40.7	4.8″
53.	$13\frac{1}{2}''$ "	42.5	5.5″
54.	16″ "	43.0	5.8″

the results presented in Table 5, show that the spatial separation between double walls does produce a very appreciable effect in increasing sound insulation.

Experiments in which a solid wood spacer replaced the alternate layers of felt and wood showed practically the same reduction, as shown

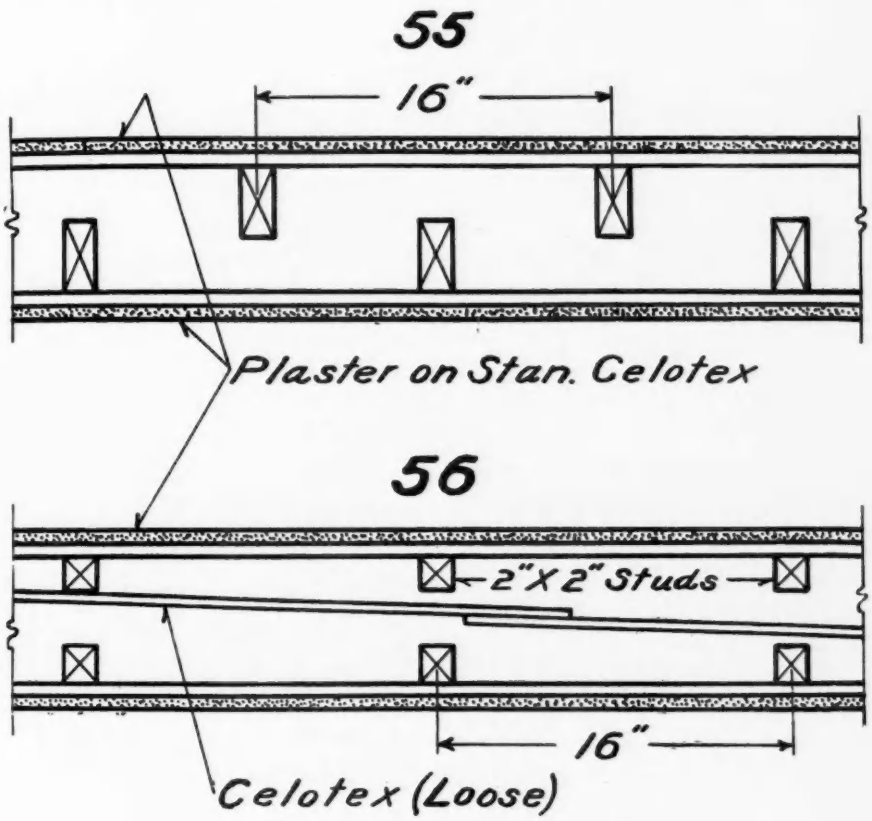
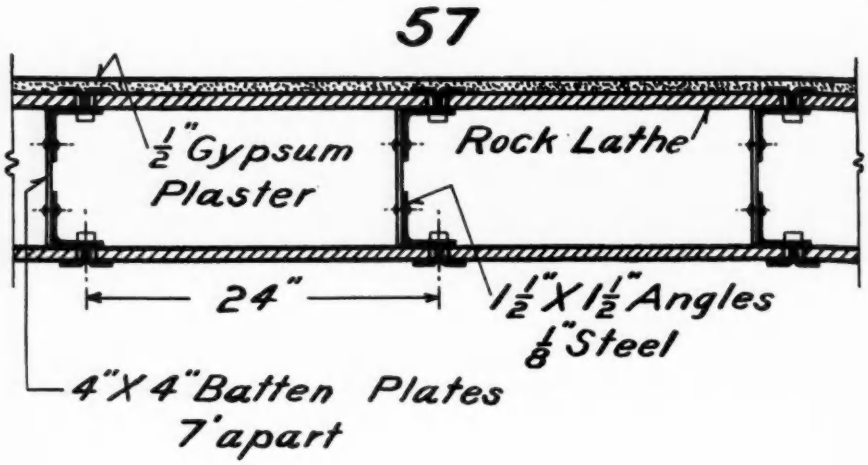
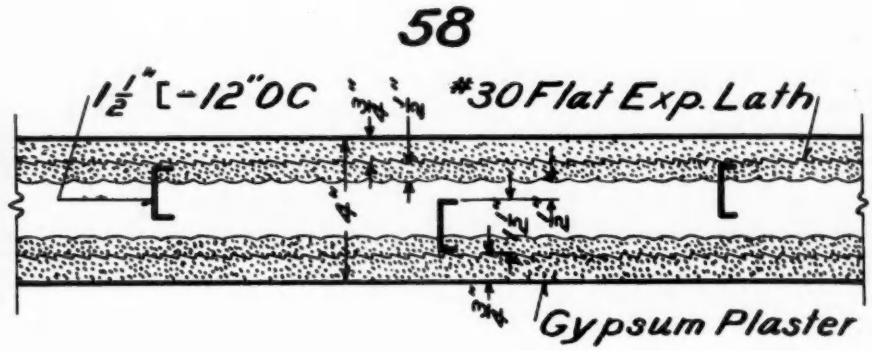


FIG. 6. *Types of double wall construction.*

by the alternate wood and felt, indicating that the increasing insulation with increasing separation is to be ascribed largely to the lower transmission across the wider air space rather than to improved insulation at the edges.

Figs. 6 and 6*a* show a number of types of double wall construction that have been tested, with results shown in Table 6.





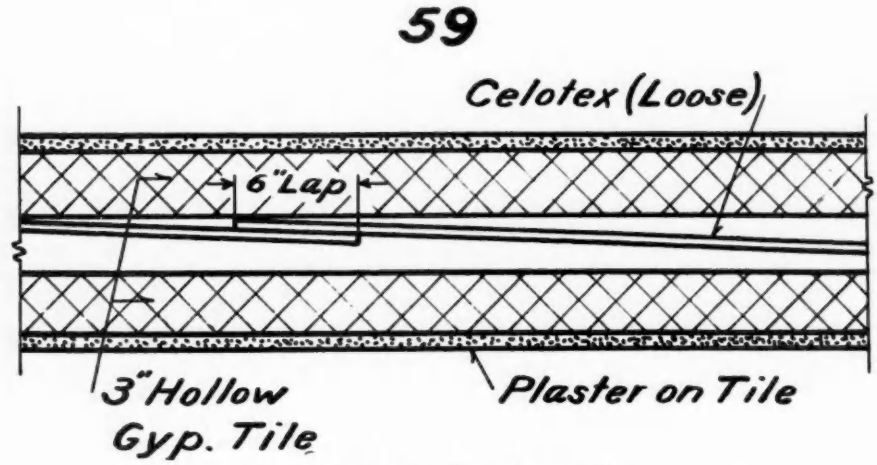


FIG. 6a. *Types of double wall construction.*

TABLE 6.

DOUBLE WALLS, CONNECTED AT THE EDGES

No.	Description	Wt/sq. ft.	Reduction	Thickness	Equiv. Masonry
55.	Staggered 2×4″ wood studs				
	(a) Metal lath, ½″ Gypsum plaster	19.8	39.1	7½″	4.0″
	(b) ½″ Celotex, ½″ Gypsum plaster	13.0	36.6	7½″	3.3″
56.	2×2″ Wood Studs, set on 6″ plate ½″ gypsum plaster on ½″ Celotex ½″ Celotex stood loosely between	12.2	46.0	8 ″	7.5″
57.	⅛″ Steel Studs, Rock lath, ½″ gypsum plaster	12.0	39.8	5¾″	4.3″
	(a) Batten plates between angles 7′ O.C.				
	(b) Batten plates 2′ O.C.	12.0	31.6	5¾″	2.3″
58.	Double metal lath on 1½″ channels, ¾″ gypsum plaster				
	(a) without cross bracing clips	18.0	43.6	4″	6.1″
	(b) with cross bracing clips	18.0	34.9	4″	2.9″
59.	Double 3″ hollow gypsum tile				
	(a) Unplastered, 3″ air space	22.0	37.5	9″	3.6″
	(b) Unplastered, 2½″ air space, ½″ felt	22.6	40.4	9″	4.5″
	(c) Same as b, with 1″ plaster	31.8	42.2	10″	5.4″
	(d) Plastered, ½″ Celotex in 2″ air space	32.0	42.1	9″	5.4″

Fig. 7 shows in a striking way the effect of the bridging by the batten plates tying the two 1/8″ steel angles forming the steel stud of No. 57 together. Table 6 indicates the limitation imposed by excessive thickness upon sound insulation by double wall construction. In only one case, that of the unclipped double metal lath construction is the double construction thinner than the equivalent masonry. The moral is that generally speaking, with structural materials one has to pay for sound insulation either in increased thickness using double construction, or by increased weight using single construction.

WOOD STUD PARTITIONS

The standard wood stud construction consists of 2″×4″ studs nailed bottom and top to 2″×4″ plate and header. One is interested to know the effect on sound insulation of the character of the plaster, whether lime or gypsum, the character of the plaster base, wood lath, metal lath, or fiber boards of various sorts, and finally of the effects of filling of

different kinds between the studs. Table 7 gives some information on these points. The plaster was intended to be standard scratch and

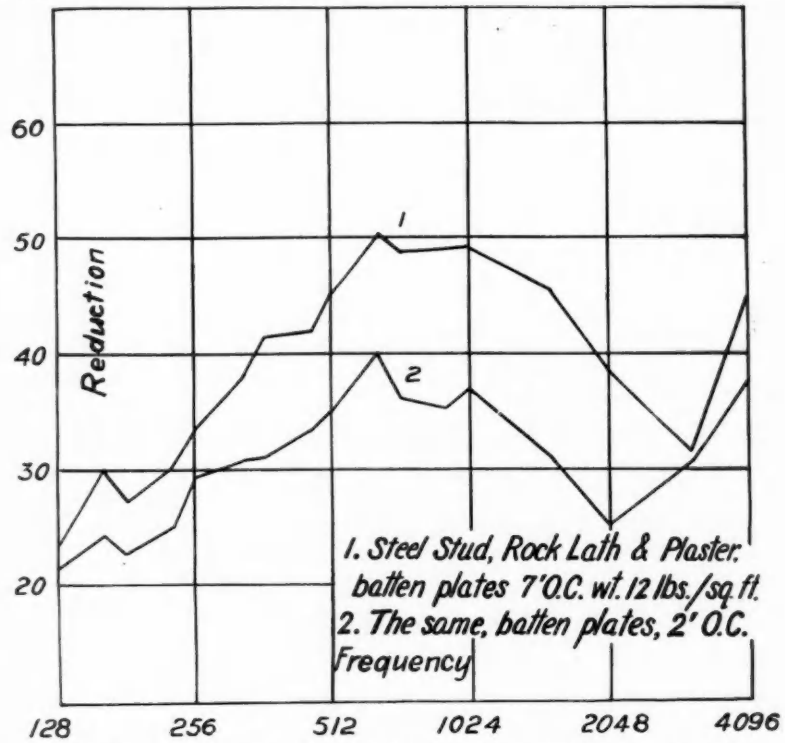


FIG. 7. *Graphs showing the effect of rigidly connecting a double wall construction.*

TABLE 7.

2×4″ WOOD STUD WALLS

Plaster Material	Plaster base	Wt/area	Reduction	Equiv. Masonry
60. Gypsum	Metal lath	17.4	29.2	1.8″
61. No. 60 filled with granulated slag	" "	27.4	36.7	3.4″
62. Gypsum	Wood lath	18.0	29.4	1.8″
63. Lime	" "	17.4	38.1	3.8″
64. None	½″ Celotex	3.0	23.1	1.0″
65. No. 64 filled with sawdust	" "	6.6	29.9	1.8″
66. Gypsum	" "	12.0	32.7	2.3″
67. No. 66 filled with sawdust	" "	15.6	35.0	2.9″
68. Gypsum, ⅝″ thick	½″ Masonite	16.0	38.0	3.8″
69. Gypsum	½″ Felt, ⅞″ furring, Metal lath	18.0	40.0	4.6″

brown coats, $\frac{1}{2}''$ to $5/8''$ total thickness. The weight in each case was determined by weighing samples taken from the wall after the tests were completed. Fig. 8 shows the effect of the sawdust fill in the celotex wall both with and without plaster. The contrast with the earlier case

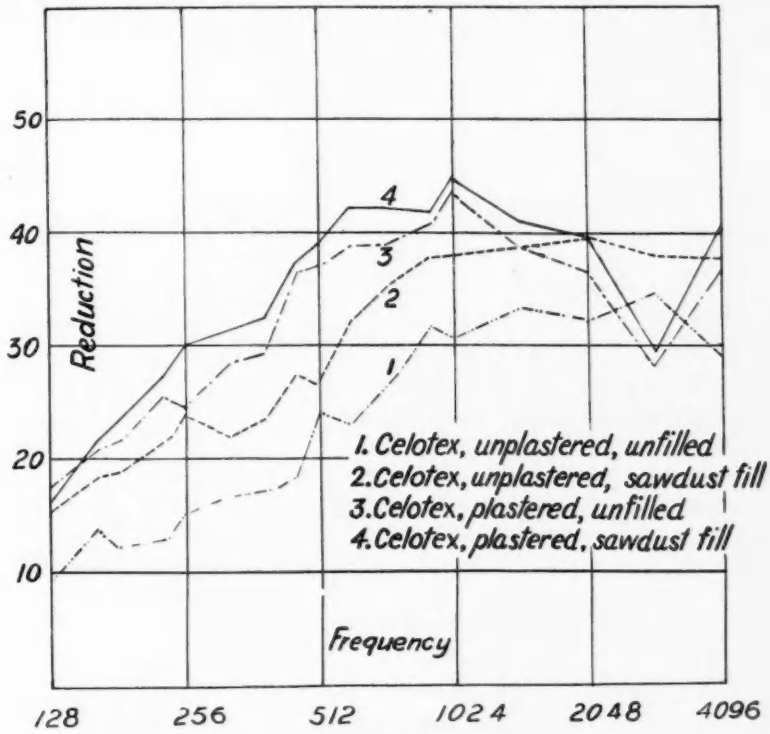


FIG. 8. *Graphs showing the effect of filling a wood stud partition.*

where there was no structural tie between the two members, and in which the sawdust filling actually *decreased* the insulation is instructive. In the wood stud construction the two faces are already completely bridged by the studs, so that the addition of the sawdust affords no added bridging effect. Its effect is therefore to add weight, and possibly to produce a damping of the structure as a whole. This suggests that the answer to the question as to whether a lagging material will improve insulation depends upon the structural conditions under which the lagging is applied. In Fig. 8, it is interesting to note the general similarity in shape of the four curves and also the fact that the addition of the sawdust makes a greater improvement in the light unplastered wall than in the heavier partition after plastering.

GENERAL CONCLUSIONS

Fig. 9 presents graphically what general conclusions seem to be warranted by the investigation so far. In the figure the vertical scale gives the reduction in decibels, and the horizontal scale, the logarithm of the weight per square foot of the partitions. The numbered points correspond to partitions described in the preceding text.

1. For continuous masonry as defined above, the reductions will fall very close to the straight line plotted. It should be emphasized that

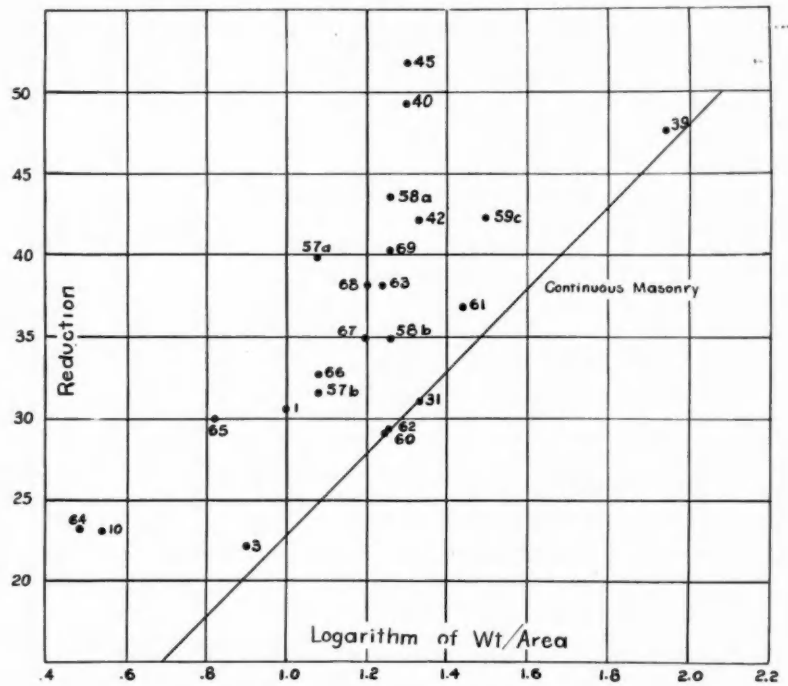


FIG. 9. *Comparison of continuous masonry with other types of construction.*

this linear relationship between the reduction and the logarithm of the weight holds only for partitions of clay and gypsum tile, solid plaster on metal lath and metal channels, and for brick. Wood stud construction with gypsum plaster falls on this line (No. 62). Lime plaster on wood studs, and gypsum plaster on fiber board plaster bases on wood studs give somewhat greater reductions than continuous masonry of equal weight, (Nos. 63, 68, 66). Glass and steel show greater reductions than masonry of equal weights, (Nos. 1 and 10).

2. The reduction afforded by double constructions is a matter of the

structural and spatial separation of the two units of the double construction, cf. 45, 40, 42, also 58a and 58b. In double constructions with only slight structural tying, lagging fills in the air space are not advantageous. In hollow construction, where filling appreciably increases the weight of the structure, filling gives increased insulation, cf. 66 and 67, and 64 and 65. In each case, the increased reduction due to the filling is about what would be expected from the increase in weight. It is fairly easy to see that since the filling material is incorporated in the wall, it can have only slight damping effect upon the vibration of the structure as a whole. Following this line of reasoning, the increase in reduction due to filling should be proportional to the logarithm of the ratio of the weight of the filled to the unfilled wall. This relationship is approximately verified in the instances cited. However, the slag filling of the metal lath and plaster wall (No. 60 and 61) produces a somewhat greater reduction than can be accounted for by the increased weight so that the character of the fill may be of some slight importance.

3. Taken altogether, the investigation shows that the practical solution of the problem of sound insulation without excessive weight lies in the direction of double constructions with a minimum of intimate contact between the separate units of construction rather than in the choice of materials that are intrinsically "sound proof." Tests have been made of numerous special types of construction proposed as sound insulating walls, some of which have proven fairly successful in practice. It is to be said that in such cases, the virtues of the finished wall are probably to be ascribed more to details of construction than to the inherent properties of the materials used.

Comparison with Bureau of Standards Findings

During the last few years, extensive investigations of this general problem have been carried on at the Bureau of Standards.[2] The method employed is decidedly different from that employed here. Qualitatively, the general conclusions drawn in the foregoing, are in agreement with the results of that work. The degree of quantitative agreement leaves much to be desired.

Table 8, gives a comparison of results by the two methods for a number of constructions for which the specifications are essentially the

[2] B. S. Sci. Paper, 526, Eckhardt and Chrisler.
B. S. Tech. Paper, 337, V. L. Chrisler.
B. S. Sci. Paper, 552, V. L. Chrisler.
B. S. Research Paper, 48, V. L. Chrisler & W. F. Snyder.

same. It will be noted that the Bureau of Standards figures are uniformly higher, on the average about 18 decibels. A part of this difference is due to the difference in the frequencies used. The Bureau of Standards tests give greater importance to frequencies above 1000 vibrations per second while the Riverbank tests give greater weight to the frequencies below 1000. The reduction for high tones is always

TABLE 8.

COMPARISON OF BUREAU OF STANDARDS WITH RIVERBANK TESTS ON SIMILAR CONSTRUCTIONS

Description	No.		Wt/Area		Reduction	
	B.S.	R.	B.S.	R.	B.S.	R.
1½″ Solid metal lath and plaster		26		13.9		26.1
2″ Solid metal lath and plaster	[1]22		18		46.2	
2″ Solid metal lath and plaster	[2]59				38.6	
2½″ Solid metal lath and plaster		33		23.2		33.0
2×4″ Wood stud, metal lath gypsum plaster	10	60	20.2	17.4	53.7	29.2
2×4″ Wood stud, wood lath gypsum plaster	4	62		18.0	46.9	29.4
2×4″ Wood stud, wood lath lime plaster	3	63		17.4	69.1	38.1
3″ Hollow Gypsum tile, 1¼″ gypsum plaster	30b	31a		22	42.0	31.4
4″ Clay Tile, 1¼″ Gypsum plaster	[1]28	36		28.6	47.8	35.6
4″ Clay Tile, 1¼″ Gypsum plaster	[3]66		29		44.0	
Double 3″ Hollow gypsum tile 1¼″ gypsum plaster	30c	59c		31.8	61.9	42.2
8″ Brick wall, 1″ Gypsum plaster	81	39	87	88	57.2	47.5

[1] B. S. Scientific Paper No. 526.
[2] B. S. Scientific Paper No. 552
[3] B. S. Research Paper No. 48.

greater than for low tones, so that the Bureau's *average* reduction would be higher, even if the two methods gave the same values at a given frequency. In the earlier paper, the Bureau gives one frequency range, in the later papers two frequency ranges below 1000, and two above 1000. The standard practice in the Riverbank tests has been to use 12 tones below and 4 tones above 1000. The more frequent occurrence in practice of sounds in the lower range was the reason for the latter choice of test frequencies. Thus only the highest soprano voices attain a frequency of 1000 vibrations per second, while only a few orchestral instruments, produce tones as high as this frequency.

In addition to this outstanding difference in results, it is also to be noted that the relative values by the two sets of tests are not the same. Thus the Riverbank tests show that the 8 inch brick wall gives the greatest reduction, while the Bureau of Standards finds that the lime plaster on wood lath on wood studs gives a reduction more than 12 dbs. greater than does the 8 inches of brick. It will be noted further that there is no close correspondence between the order of the reduction shown by the two series of tests. This lack of correspondence may also be ascribed in part to the difference in the choice of test tones. There are doubtless other factors, fuller knowledge of which it is to be hoped will reconcile these differences.

This lack of agreement should not discredit laboratory methods as one means of acquiring useful information upon the important problem of sound insulation. It should emphasize the fact that there are inherent difficulties in making laboratory tests that shall be of general significance due to the large number of factors, some doubtless still unrecognized, that enter into the transmission of sound by way of intervening walls. What has been aimed at has not been so much the relative merits of different materials and constructions as to find out the direction which normal construction should take to give increased sound insulation.

COEFFICIENT OF TRANSMISSION OF SOUND

By F. R. Watson
University of Illinois

Introduction

When sound impinges on a layer of material, such as hairfelt, part of the incident energy is reflected while the remainder penetrates the material and is either absorbed or transmitted. (See Fig. 1) The amount reflected from hairfelt is small, but the absorption in the porous channels

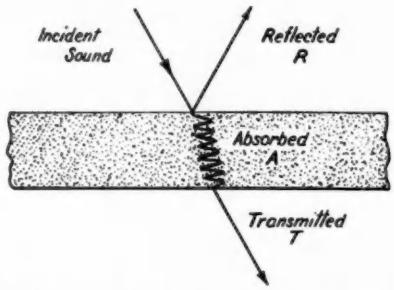


Fig. 1. *Reflection, Absorption and Transmission of Sound.*

may be large, depending on the thickness. If, however, sound is incident on a solid partition such as a plastered tile wall, 95 per cent or more of the sound is reflected; and very little is absorbed. Any transmitted sound for this case is due almost entirely to a vibration of the partition under the action of the incident sound, whereby vibrations (sound waves) are set up in the air on the side opposite the incident waves.

Experimental

In an investigation conducted a number of years ago by the writer,[1] the transmission and reflection of sound by various materials were measured. An adjustable metal organ pipe generated the sound which was directed by a parabolic reflector to the material under test, where it suffered reflection and transmission. Rayleigh disc resonators measured the relative intensities of the transmitted and reflected sound.

The results expected for normal action of sound are pictured in Fig. 3, which shows the measurements obtained for hairfelt. The transmission decreases with an increase in the number of layers of material; the ab-

[1] Sound-Proof Partitions. Univ. of Illinois Engineering Experiment Station Bulletin 127, 1922.

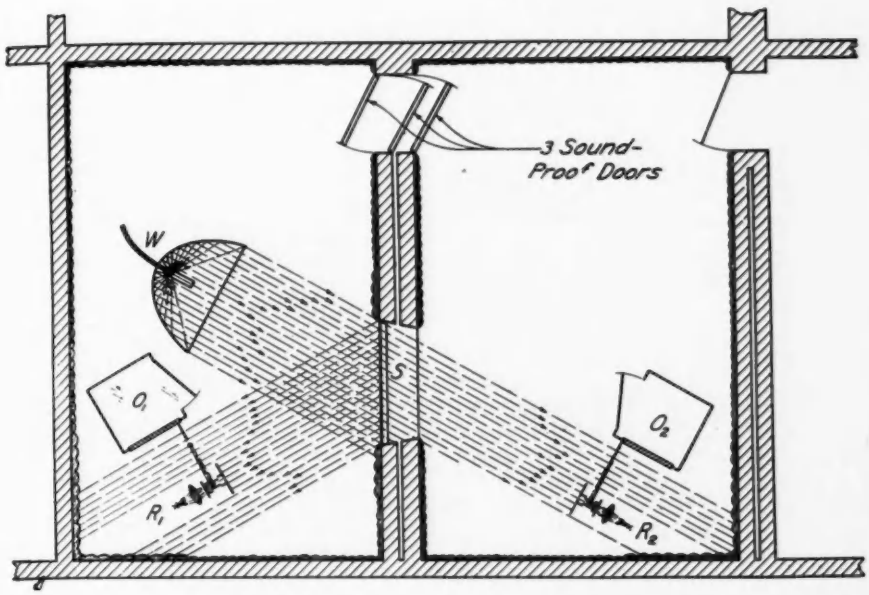


FIG. 2. *Diagram of Apparatus for Measuring the Reflection and Transmission of Sound.*

sorption and reflection increase with the thickness, the latter approaching a constant value.

A surprising result was obtained with paper-lined hairfelt, when it was found that more sound was transmitted by two layers than by one

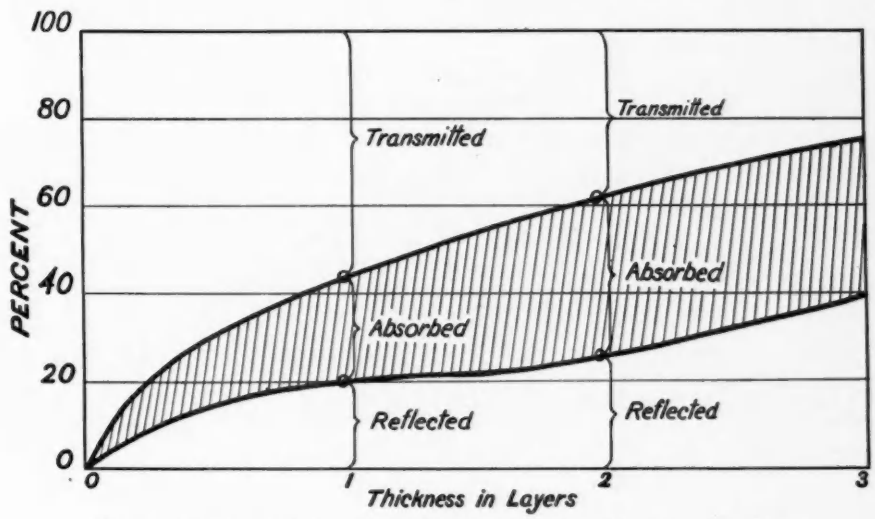


FIG. 3. *The Transmission, Absorption and Reflection of Sound by Hairfelt.*

layer. This puzzling action was explained when the values for both transmission and reflection were plotted, as shown in Fig. 4. For one layer, a considerable reflection took place, with a correspondingly small amount of transmitted sound. For two layers, however, much more sound was transmitted than by one layer; but the reflection decreased. This action was set up by a resonant vibration of the partition that

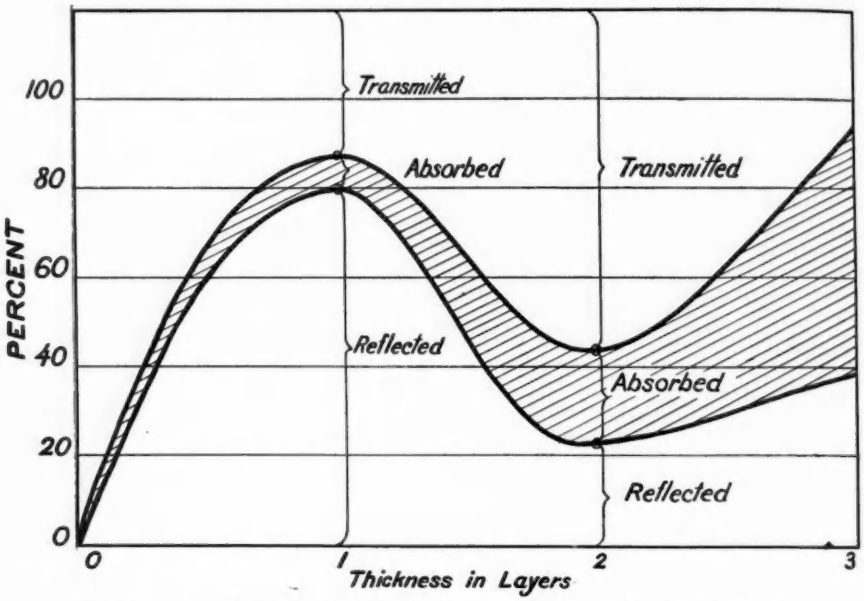


FIG. 4. *Action of a Vibrating Partition in Transmitting and Reflecting Sound.*

created a considerable volume of sound on the side opposite the incident waves.

Later experiments on plaster partitions showed that sound was stopped more effectively by these heavy, stiff constructions than by the pliable, porous hairfelt. Nearly all of the sound striking the plaster walls was reflected.

At the present time, the writer is starting a new series of transmission measurements, but with modified apparatus. Sound is generated when an electrodynamic loud speaker is operated by current from an audible frequency audion oscillator. By simple adjustment of capacities and inductances, a practically pure alternating current can be obtained for any frequency from 60 to 80,000 per second, with a variation of intensity of 2000 fold or more without change in frequency. The current is

amplified by a three-stage amplifier and led through low pass electric filters to remove overtones. The output current is measured by a thermocouple and galvanometer and reaches the electrodynamic speaker through an adjustable audibility meter. A motor driven air condenser

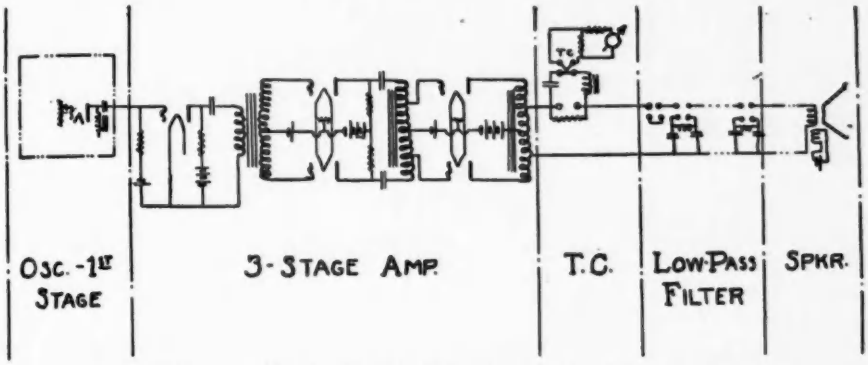


FIG. 5. *Diagram Showing Electric Apparatus Used for Generating Sound.*

with rotating blades, connected to the audion oscillator, allows the pitch of the resultant sound to be varied above and below a selected value. This variation of pitch (and wavelength) serves efficiently to break up any fixed pattern of sound in the test room.

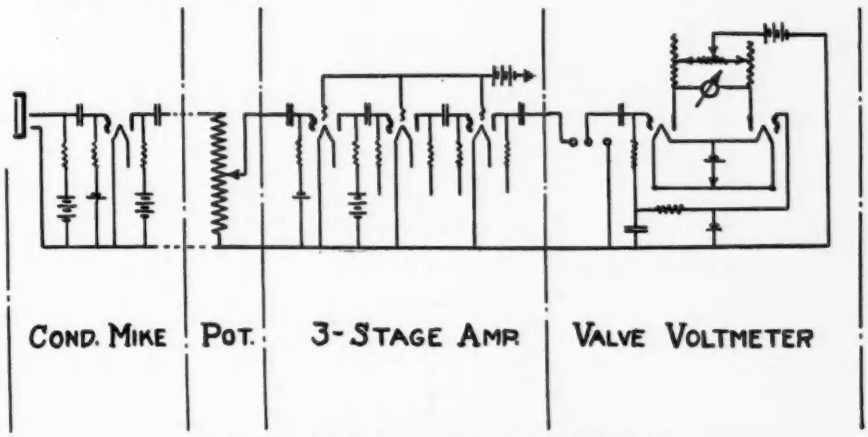


FIG. 6. *Electric Apparatus for Receiving Sound.*

When sound from the source transmits a partition, it is measured by a specially developed electric receiving apparatus. A condenser microphone picks up the sound, the resulting electric current passing through a potentiometer, after which it is amplified and measured by a Wynne-Williams balanced tube valve voltmeter.

THEORY OF SOUND TRANSMISSION

Usually observers measure the intensities of the incident sound, I, and the transmitted sound, I_1; and take the ratio I/I_1, which is called the "reduction factor," sometimes expressed as the logarithm I/I_1. This reduction factor does not directly give the coefficient of transmission per unit area. Davis has shown[2] that the sound E from the source is lost partly by absorption at the walls of the room and partly by the transmission into the test room, where some sound is transmitted back

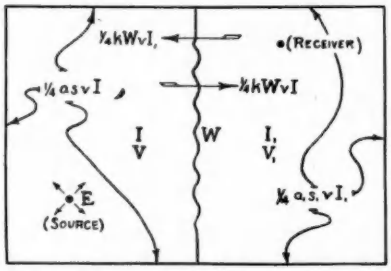


FIG. 7. *Diagram Showing the Action of Sound in Two Adjacent Rooms.*

into the source room. He develops the equation: $I_1/I = kw/a_1s_1$, where I_1 and I are the respective intensities in the test and source room, k is the coefficient of transmission of sound per unit area, w the area of the dividing partition, and a_1s_1 the absorption of the surfaces in the test room.

The factor k is the quantity to be found. Experimentally several procedures are suggested:

1. If the absorption a_1s_1 is changed by definite amounts, the ratio I_1/I may be plotted against $1/a_1s_1$, so that

$$I_1/I \div 1/a_1s_1 = kw$$

from which k may be calculated.

2. If an opening between the source and test room is covered successively by similar layers of a material like hairfelt, $k = k_1^n$ where k_1 is the transmission through one layer, and n is the number of layers;

$$a_1s_1 = A_1S_1 \text{ (absorption of surfaces except the partition)}$$
$$+ aw \text{ (absorption by partition)}$$
$$+ kw \text{ (transmitted by partition)}$$

[2] Phil. Mag., 50, 75, 1925; 2, 543, 1926.

$$=A_1S_1\left[1+\frac{w}{A_1S_1}(a+k)\right].$$

The equation may then be written:

$$\log I_1/I=n\log k_1+\log w-\log A_1S_1-\log\left[1+\frac{w}{A_1S_1}(a+k)\right].$$

The term

$$\log\left[1+\frac{w}{A_1S_1}(a+k)\right]=\frac{w}{A_1S_1}(a+k)-\frac{1}{2}\frac{w^2}{(A_1S_1)^2}+\frac{1}{3}\frac{w^3}{(A_1S_1)^3}\quad\text{etc}$$

if $-1<w/A_1S_1(a+k)<+1$. This condition is fulfilled if A_1, the average coefficient of absorption of the room is large; since $(a+k)=1-r$ where r is the reflection coefficient, and $(a+k)$ is thus a fraction. The equation then becomes approximately:

$$\log I_1/I=n\log k_1+\log w-\log A_1S_1-\frac{w}{A_1S_1}(a+k).$$

Experiments conducted in this way give approximately a straight line when $\log I_1/I$ is plotted against n, so that $\log k=(\log I_1/I)/n$.

The *vibra ion* of the partition depends on the mass, stiffness and viscosity of the partition. For instance, the displacement may be shown[3] to be: $q=Q/a(n^2-p^2)$ where q is the displacement of the partition, Q is the impressed force acting, n and p represent the frequencies respectively of the partition and the impressed force, and a is the coefficient of inertia (mass). If the frequency of the impressed force is much greater than the natural vibration of the partition, n^2 may be neglected and $q=-Q/p^2a$, showing that the displacement depends only on the mass of the partition,—the greater the mass, the smaller the displacement. This is the usual case in buildings, where the natural vibrations of partitions do not exceed about 50 per second, while the impressed sounds vary from about 128 to 5000 vibrations per second. If the frequency of the impressed force is quite small compared with that of the partition p^2 may be neglected and the equation becomes: $q=Q/n^2a=Q/c$, where $n^2=c/a$, and c is the coefficient of stiffness. In this case the stiffness of the partition determines the amount of dis-

[3] Lamb, Dynamical Theory of Sound, Sec. 9.

placement. Such a case would follow if a partition vibrates in segments of frequency considerably greater than that of the incident sound.

CONCLUSION

For comparison of results among different observers, it is desirable to obtain values for the constant k, the coefficient of transmission. The reduction factor I/I_1 evidently will vary with the rooms in which the tests are conducted. For solid partitions, it appears that the transmission is due almost entirely to vibrations of the structure, whereas in porous partitions, there is a large transmission of the wave motion.

AN AUDIOMETRIC METHOD FOR MEASURING SOUND INSULATION

By Wallace Waterfall
The Celotex Co,

Various methods are in use for measuring the sound insulating properties of partitions. Most of the data available have been obtained on test panels built in a wall opening in a specially prepared laboratory. Quiet conditions are essential to accurate results.

In commercial work it is frequently necessary to make sound insulation measurements in buildings where noise conditions cannot be controlled. The equipment used must be easily portable. This paper reports an audiometric method of measuring sound insulation which has been found to be satisfactory in practice and to be in agreement with sound insulation measurements made by the reverberation method and by a method in which the relative sound intensities on opposite sides of a partition are measured instrumentally.

A diagram of the equipment used in the audiometric measurements is given in Figure 1. The beat-frequency oscillator furnishes essentially pure tones of any desired frequency. These are amplified by a two-stage battery operated amplifier. The output of the amplifier is fed into a motor-driven double-pole double-throw switch which connects it alter-

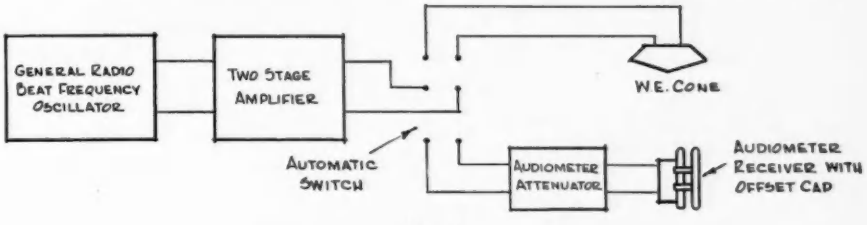


Fig. 1.

nately to a Western Electric cone and to the attenuator and receiver of the standard 3A Audiometer. In this way the tone is audible alternately through the loud speaker and through the audiometer receiver. The rate of alternation depends somewhat upon the reverberation in the source room, as it is necessary to allow sufficient time for the sound to build up to its maximum loudness. A rate of approximately 20 alternations per minute was used in the investigations reported here. This gave a duration of sound through the loud speaker of approximately 1.5 seconds.

The method of measurement consists of placing the loud speaker on one side of the partition at a distance usually six feet or more from the partition and then, by adjusting the attenuator, making loudness matches first on one side and then on the other side of the partition at a distance of approximately three feet from it. The loudness match is made for all of the tones on one side of the partition before the observer moves to the other side. The attenuator is graduated in decibels so the difference in the readings on the two sides gives directly the reduction factor of the partition for the tone used.

At the beginning of the investigation it was necessary to decide upon how many tones should be used. It is known that a test at one frequency may not show the relative values of partitions. In determining the tones to use, tests made by Dr. Paul E. Sabine for The Celotex Company were studied. These tests were made by the reverberation method. Each partition was tested for seventeen tones in the frequency range from 136 to 4096 cycles. Figure 2 shows the reduction factors in decibels for 256, 512, 1024 and 2048 cycles, along with the averages for these four

TESTS BY P. E. SABINE, RIVERBANK LABORATORIES
REDUCTION OF PARTITIONS IN DECIBELS

Partition No.	1	2	3	4	5
256 cycles	22.6	27.1	30.6	32.8	38.6
512 cycles	34.0	35.5	40.7	39.6	49.5
1024 cycles	40.8	42.1	47.1	38.2	51.3
2048 cycles	36.6	39.6	48.7	33.4	46.3
Average	33.5	36.1	41.8	38.0	46.4
Av. for 17 tones from 136 to 4096	30.3	32.1	39.0	34.2	43.2
Difference	3.2	4.0	2.8	3.8	3.2

FIG. 2.

tones and the averages for the seventeen tones. It will be noticed that the averages for the four tones chosen are greater than the averages of the seventeen tones. This is due to the fact that the majority of the seventeen tones were of low frequency. The difference between the averages, however, varies only from 2.8 to 4.0 decibels. The average variation for the five partitions is 3.4 decibels and the variation from this average is less than 1 decibel. It appears therefore that tests can be made for the four selected frequencies which should give relative values

on various partitions practically like the values obtained from seventeen tones. These four tones were therefore selected for use.

The audiometric method was first tried in measuring the sound insulation value of fifteen partitions of different construction erected in a large apartment building. These partitions were approximately fifteen

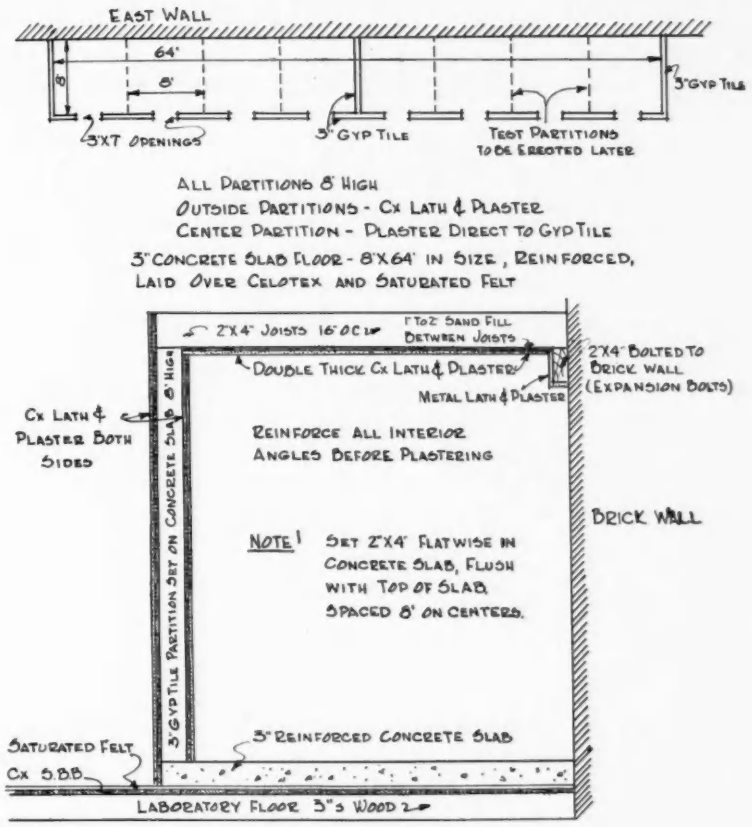

EAST WALL
64'
8'
8'
3"XT OPENINGS
3" GYP TILE
TEST PARTITIONS TO BE ERECTED LATER
3"GYP TILE
ALL PARTITIONS 8' HIGH
OUTSIDE PARTITIONS - Cx LATH & PLASTER
CENTER PARTITION - PLASTER DIRECT TO GYP TILE
3" CONCRETE SLAB FLOOR - 8'X64' IN SIZE, REINFORCED, LAID OVER CELOTEX AND SATURATED FELT
2"X4" JOISTS 16" O.C.
F TO 2' SAND FILL BETWEEN JOISTS
DOUBLE THICK Cx LATH & PLASTER
2"X4" BOLTED TO BRICK WALL (EXPANSION BOLTS)
METAL LATH & PLASTER
Cx LATH & PLASTER BOTH SIDES
REINFORCE ALL INTERIOR ANGLES BEFORE PLASTERING
BRICK WALL
NOTE! SET 2"X4" FLATWISE IN CONCRETE SLAB, FLUSH WITH TOP OF SLAB, SPACED 8' ON CENTERS.
3" GYP TILE PARTITION SET ON CONCRETE SLAB 8' HIGH
3" REINFORCED CONCRETE SLAB
SATURATED FELT
Cx S.B.B.
LABORATORY FLOOR 3"s WOOD 2"

FIG. 3.

feet long and nine feet high. In addition to testing for the four pure tones, tests were also made in each case using the tone of the 3A Audiometer. When the audiometer was used as a source the two-stage amplifier was omitted but the arrangement was otherwise the same as shown in Figure 1. The results of this complete investigation are not given here. There are several points of interest, however. Two observers (Professor F. R. Watson and myself) made all the measurements. Each observer measured independently and without knowledge of the other observer's

results. The readings were repeated on each partition until the observers agreed within 4 decibels on the average reduction for the four tones. The average difference between the observers for the fifteen partitions was only 2.7 decibels. The reduction for the 3A Audiometer tone bore no fixed relation to the pure tone average. For some partitions the two were equal. The 3A tone reduction was never lower than the other and for one partition it was 13 decibels higher. Usually the reduction for the 3A tone was from 3 to 7 decibels greater than the average reduction for the four pure tones.

Following the apartment house measurements a group of special test rooms were erected as shown in Figure 3. The rooms are so arranged that seven partitions, approximately eight feet by eight feet in dimensions, separate adjoining rooms. The test rooms are therefore eight-foot cubes with concrete floor, plaster ceiling loaded with sand, and walls of either plaster or brick. There is a single $3' \times 7'$ door opening to each room. During the tests the door openings of the source and receiving rooms were tightly closed by $4' \times 8'$ panels of 4-ply Celotex held tightly in place against the outside of the opening. Felt pads were provided to insure a tight fit. Several measurements of the reduction factor of these doors were made during the tests. The average for the 3A tone was 36 decibels and for the four pure tones was 30 decibels. The insulation value of the two doors was therefore considerably in excess of the value of any partition. To compensate for the absorption of the observer in the source room the standard practice was used of placing eight square feet of material having 70% absorption on the floor of the source room during the observers' absence.

TYPICAL AUDIOMETRIC TEST DATA
Reduction factors on wood stud, Celotex and plaster partition

Date	3A Tone	256	512	1024	2048	Average
10- 8-29	35	29	38	39	47	38
	35	29	38	43	49	40
10-18-29	41	31	35	35	41	36
	43	26	34	37	42	35
10-23-29	35	26	33	33	38	33
	35	25	26	37	40	32
11-19-29		38	32	37	38	36
Average	37	29	34	37	42	36

FIG. 4.

In this group of eight rooms measurements have already been made on seventeen types of partitions.

A typical set of test data for a standard type of partition is shown in Figure 4. All of the measurements were made by the writer. The sensation level of each tone in the source room was not kept constant during all the tests. The variation in level was necessary to avoid memory effects. The maximum sensation level was approximately 88 decibels. The data collected furnish evidence that the reduction factor of a partition is practically independent of the intensity of the incident sound if other conditions remain the same. A typical group of measurements on a partition at four sensation levels covering a range of approximately 18 decibels gave average reduction factors of 38, 38, 39 and 39 decibels respectively.

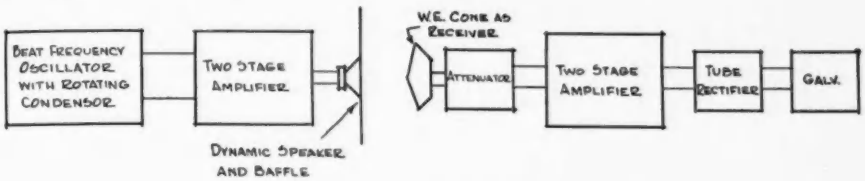


FIG. 5.

In addition to the audiometric method of measurement it was decided desirable to use some method which would be independent of the ear. During the tests it was found that some experience in making loudness matches is necessary for consistent results with the audiometric method and that even an experienced observer may show some variation from day to day. Therefore, an instrumental arrangement, such as shown in Figure 5, was set up for measuring the reduction factors of some of the same partitions which had been measured by the audiometric method. The beat-frequency oscillator was provided with an external rotating condenser which provided tones of wavering frequency. The oscillator was connected to a two stage power amplifier, different from the one used in the audiometric measurements, which in turn was connected to a dynamic speaker with a four foot square baffle. One reason for the change in amplifiers and speakers was to increase the loudness in the source room to come within the sensitivity of the receiving equipment. For the receiver several types of telephone receivers were tried. Most uniform sensitivity was obtained by using the Western Electric cone as a receiver so all measurements were made with it. The receiving equipment has been calibrated by calculation from the potentiometer in-

cluded as a part of the vacuum tube volt-meter. This calculated calibration has been checked by measurements against a Rayleigh disc and also by measurements using the calibrated 3A Audiometer as a source. It is believed that the calibration is sufficiently accurate for these sound insulation measurements.

In making the measurements the dynamic speaker was placed in one corner of the source room and the receiver was hung in the center of the source room and then in the center of the receiving room and the sound intensity measured in each. In the instrumental measurements, therefore, the receiver was at a distance of four feet from the partition. Audiometric measurements were made at four feet from the partition, as well as three feet to see whether this slight difference would affect the results. In a large absorbent room the distance from a partition will undoubtedly effect the measured reduction but in these small reverberant rooms the difference between three and four feet cannot be detected.

Measurements had first been tried using a constant rather than a wavering tone and swinging the receiver. This was not found to be satisfactory as it was impossible to repeat measurements. The wavering tone was finally adopted with the receiver hanging in a stationary position. The tone was varied at the rate of approximately 150 times per minute. This made the galvanometer needle sufficiently steady for reading.

Instrumental measurements have now been made on seven partitions, all of which had previously been tested by the audiometric method. It happens that these seven partitions all had average reduction factors between 33 decibels and 39 decibels. For five of the partitions the difference between the average values determined by the instrumental and audiometric methods was 1 decibel or less. For the other two partitions the difference was 2 decibels.

REDUCTION FACTORS BY THREE METHODS OF MEASUREMENT
2″x 4″ wood studs 16″ O.C., Celotex and plaster both sides—Reduction in decibels

Method	256	512	1024	2048	Average
Reverberation (P. E. Sabine)	22.6	34	40.8	36.6	33.5
Audiometric	29	34	37	42	36
	250–330	420–600	850–1150	1500–2300	
Instrumental	29	30	36	43	35

FIG. 6.

A more detailed comparison of the results between the audiometric, instrumental and reverberation methods is interesting. Figure 6 contains the values by these three methods obtained on a standard type of partition constructed of 2″ x 4″ wood studs, 16″ on centers, with Celotex and ½″ plaster on both sides. The partition tested at Riverbank Laboratories was built in a 6′ x 8′ opening and the weight per square foot was 11.5 pounds. The results show surprisingly good agreement for the three methods.

During the tests there were a number of interesting observations, a few of which are reported in the following paragraphs.

One of the partitions which was erected in the apartment house was also constructed between two test rooms in the laboratory. The partitions are of different size and it is interesting to compare the results, tabulated in Figure 7. From this data it seems safe to conclude that the reduction factors on partitions eight feet square can be taken as

EFFECT OF SIZE OF PARTITION ON REDUCTION FACTOR
AUDIOMETRIC MEASUREMENTS—REDUCTION IN DECIBELS
3″ Gypsum tile partition with Celotex on both sides, plastered

Size of Partition	3A Tone	256	512	1024	2048	Average
9′-0″×15′-0″	49	34	39	42	42	39
8′-0″× 8′-0″	49	29	35	41	44	37

FIG. 7.

representative of partitions of larger size. No data is yet available to show how small a panel can be and still give representative values. This should depend to some extent upon the construction of the partition.

The 3A Audiometer tone is supposed to be representative of speech and, since there was considerable difference between the audiometer and tone reduction factors in some cases, an attempt was made to measure directly the reduction factors for reproduced speech and music by the audiometric method. Sousa's "Stars and Stripes Forever" and a "Sam and Henry" talking record were used. It was found very difficult to make the measurements but the values obtained seem to be more nearly in agreement with the average reduction for the four pure tones than with the audiometer reduction factor. To illustrate this, the music and speech each gave a reduction factor of 32 decibels on a partition where the average reduction for the four tones was 33 decibels and the reduction for the 3A tone was 43 decibels.

An attempt was also made to determine the effect of different reduction factors on the intelligibility of speech. Standard word articulation lists, such as described on page 263 of "Speech and Hearing," by Harvey Fletcher, were used. Three partitions having average reduction factors of 25 decibels, 36 decibels and 39 decibels respectively were tried and gave articulation values of 17%, 10% and 6%. These results are not particularly significant because not enough measurements were made, but it is believed that articulation measurements might be used to determine what constitutes a satisfactory partition under a definite set of conditions.

Several measurements were made to determine the effect of absorption in the receiving room on the reduction factor. In the instrumental measurements the effect was very definite. One hundred and twenty square feet of a material having an absorption coefficient of .47 at 512 cycles was placed on the floor and against the walls of the receiving room and the average reduction factor for the four tones was increased by 8 decibels. The effect of the absorption was not so definite in the audiometric measurements but in almost all cases the measured reduction increased with absorption. Such a result is to be expected and in applying sound insulation test data to practical problems this effect should be carefully considered.

DECEMBER 9, 1929.

REVERBERATION TIME IN "DEAD" ROOMS

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Introduction

With the advent of radio broadcasting and sound pictures very "dead" rooms have been built, and the significant problem of just how much reverberation should be used in broadcasting and recording presents itself. The direct measurement of reverberation time or its calculation by the aid of a reliable formula, then, is an important aspect of applied acoustics. A reverberation time formula enables one to calculate the reverberation time once the volume, surface area and average absorption coefficient of the surface of the room are known; or if the reverberation time is measured it enables one to calculate the average coefficient of absorption of the surface treatment. A correct reverberation time formula is, therefore, much to be desired.

Theories of reverberation leading to Sabine's reverberation time equation have been given by W. C. Sabine (1900),[1] Franklin (1903),[2] Jaeger (1911),[3] Buckingham (1925).[4] Recently Schuster and Waetzmann (1929)[5] have pointed out that Sabine's formula is essentially a "live" room formula and they have shown as we also show that the reverberation time equation varies somewhat with the shape of the room. The present paper presents an analysis based on the assumption that image sources may replace the walls of a room in calculating the rate of decay of sound intensity after the sound source is cut off, which gives a form of reverberation time equation more general than Sabine's; it points out the difference between the basic assumptions leading to the two types of formulae; it adds experimental data which support the more general type; and it ends with the conclusion that no one formula without modification is essentially all inclusive.

Reverberation Time Formulae

Sabine's Formula. If sound is emitted at a constant rate in a room, the sound energy density will build up till an equilibrium is reached

[1] W. C. Sabine, Collected Papers on Acoustics.
[2] Franklin, Phys. Rev. 16, 372 1903.
[3] Jaeger, Wiener Akad, Ber., Math.-Naturw, Klasse, Bd. 120 Abt. IIa, 1911.
[4] Buckingham, Bur. Standards, Sci. Paper, No. 506, 1925.
[5] Schuster and Waetzmann, Ann. d. Phys. March 1929; also Textbook by Muller-Pouillet, Vol. on Acoustics Chapter VII, pp. 456–460.

between the energy emitted and the energy absorbed. When the source is turned off, this energy density will drop off at a rate depending on the absorbing power of the walls and fixtures. Early in his researches Sabine found "the general applicability of the hyperbolic law of inverse proportionality" between reverberation time and absorbing power as given by the relation

$$T=\frac{KV}{a} \tag{1}$$

where T is the duration of residual sound, V the volume of the room, and a the absorbing power of the walls. Standardizing reverberation time as the time required for the intensity of sound to drop to one millionth of its value, Sabine determined the constant K, and obtained in English units,

$$T=\frac{0.05V}{a}=\frac{0.05V}{S\alpha_a} \tag{2}$$

where S is the surface of the room and α_a is the average coefficient of absorption defined by the relation

$$\alpha_a=\frac{s_1\alpha_1+s_2\alpha_2+s_3\alpha_3+\cdots}{s_1+s_2+s_3+\cdots}=\frac{\sum s\alpha}{S} \tag{3}$$

where $s_1, \alpha_1, s_2, \alpha_2$, etc., are the elements of surface and the corresponding absorption coefficients, and S is the total surface. This method of averaging assigns equal weights to the elements of surface, which means that it tacitly assumes a perfectly diffuse condition of the energy density. If, on the other hand, an ordered condition exists, proper weights will need to be assigned to the elements of surface.

Although Sabine tested his reverberation time formula for rooms of various shapes and volumes, ranging from a "small committee room to a theatre having a seating capacity of nearly fifteen hundred," he did not increase the absorption power to the extent that the rooms became "dead," the majority of reverberation times ranging from 4 sec. to 1.5 sec. Thus Sabine did not test his formula in "dead" rooms and no doubt he did not expect it to have a meaning for the extreme case when the average coefficient is unity, for under this condition there can be no hang over of sound and hence no reverberation time, except as one wrongly calls the time for sound to travel from the source to the observer a reverberation time. Yet formula (2) does not become zero but

simply reduces to $T=0.05V/S$ for this extreme case of absorption, and the reverberation time, strange as it may seem, becomes a function of the shape of the room, and reduces to zero only when S becomes very much greater than V. What meaning the equation has for the out-of-doors is certainly very vague. It is evident that the formula fails for this extreme case, and it seems natural to test its validity also for very "dead" rooms.

Recent experiments in the Sound Stage, Sound Picture Laboratory, Bell Telephone Laboratories, indicate the failure of Sabine's formula when the average absorption coefficient for a room is rather high—above 0.5. The results are recorded in another part of this paper.

The More General Formula. Although Sabine's formula had its beginning in the experimental study of "live" rooms, it is also derivable from a theorectical study. A careful analysis of the theoretical basis of reverberation theory is, therefore, important in the development of a new formula.

This necessary analysis is aided by the method of images. Just as a plane mirror produces an image of a source of light, so also will a reflecting wall with dimensions large as compared with the wave length of the sound wave produce the image of a source of sound. An image will be produced at each reflection. In a rectangular room, the source images will be discretely located through space. This infinity of image sources may replace the walls of the room, for they will produce an energy density at a point in the room just as if they were absent and the walls were present.

One may picture the building up of the sound as follows. As soon as the source is turned on the infinity of image sources are at that instant all turned on. The walls are imagined removed, and hence at a given place in the former enclosure sound energy will begin to arrive first from the source, then from the first reflection image sources, then from the second reflection image sources, and so on till the energy arrives from the most distant sources.

The decay of sound may be pictured thus. When the source is stopped one may imagine that all the image sources are simultaneously stopped. The first drop will be heralded by the direct wave, then a series of drops by the waves from the first reflection image sources, then drops from the second reflection image sources and so on, and so on, till the contribution of all the remaining image sources is not sufficient to affect the hearing. Thus the decay ends. The effect of all the image

sources is a million times greater than the effect of all those located *beyond* a distance cT; this is the meaning of reverberation time T in the new picture, c being the speed of sound in air.

We wish to emphasize that this picture which gives the details of the decay, involving as it does the geometry of the room, the distribution of the absorbing material, the discontinuous nature of the energy decay, and interference phenomena, is fundamental in developing reverberation theory. Just how many details we shall represent by averages depends upon the degree of simplification desired. We shall ignore interference phenomena in the following considerations, and accordingly we shall attempt to make reverberation time meter measurements independent of this effect. We shall, following Sabine's experimental results, assume that α_a defined by equation (3) may be considered as the uniform absorption coefficient of the walls; yet we realize that for an ordered condition of the sound waves this procedure may need to be modified. We shall apply the method of images first to a few special cases involving rooms of simple geometrical form in order that later we may with better understanding approach the general case.

Consider a spherical room of diameter D and volume V, with uniform absorption over the surface and with a sound source emitting $\overline{E}$ energy units per second located at its center. In a time $D/2c$ sec. after the source is turned on, the sphere will be filled with energy directly from the source of average density $\overline{E}D/2cV$, and then a first reflection will begin. The reflected wave will travel back to the center, then spread out and in a time D/c sec. after the reflection first started a second reflection will begin. The average density due to the first reflected energy then is $\overline{E}DR/Vc$ where R is the coefficient of reflection defined by the relation

$$R=(1-\alpha_a). \tag{4}$$

After another D/c sec. interval, the third reflection will begin and the average energy density of the second reflected energy is, $\overline{E}DR^2/Vc$. As time proceeds, at the end of each D/c interval, a new reflection begins and a new increment will have been added to the total average density of the enclosure. Finally equilibrium is established and the total average energy density is given by the infinite sum

$$S_\infty=\frac{\overline{E}D}{Vc}\left(\frac{1}{2}+R+R^2+R^3+\cdots R^n+\cdots\right). \tag{5}$$

When the source is turned off, the decay begins. At a point of observa-

tion near the center the energy directly from the source stops, then in D/c seconds the average energy represented by the second term of equation (5) vanishes; in D/c seconds more the third term vanishes, etc., etc. The total average energy at any D/c interval during the decay will be given by the equation

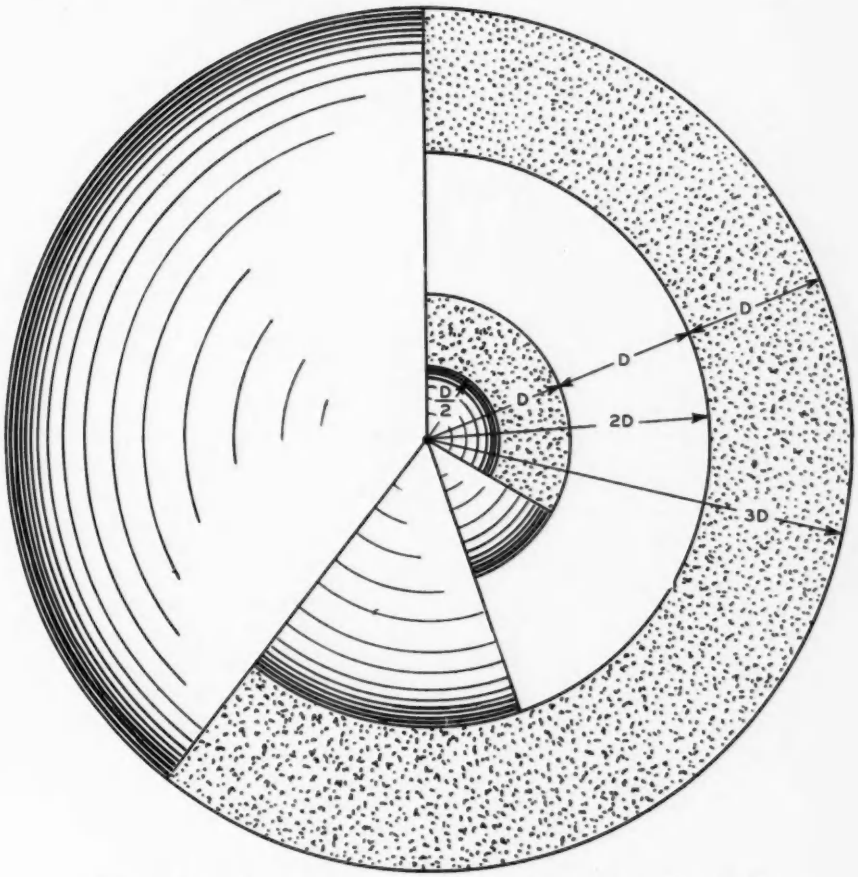

FIG. 1. *Illustrating a source at the center of a spherical room and a series of spherical concentric image sheets.*

$$S_\infty - S_n = \frac{\overline{E}DR^n}{Vc(1-R)} = \frac{6\overline{E}}{S\alpha_a c}\, e^{\frac{cS[\log_e(1-\alpha_a)]t}{6V}} \tag{6}$$

remembering that $n = ct/D$, that $R = (1-\alpha_a)$, and that for a sphere $D = 6V/S$. We must keep in mind that t is not a continuous function but given by, $t = nD/c$ where n is an integer, and also that the absorbing material is assumed uniformly distributed over the surface.

One may interpret the process just described by the aid of the method of images. The walls are imagined removed and image sources are substituted. These images are imagined to be spherical concentric sheets. (See Figure 1). They are separated by a distance D, the distance between two successive reflections. The first image sheet contributes to the enclosure the average energy density given by the second term of

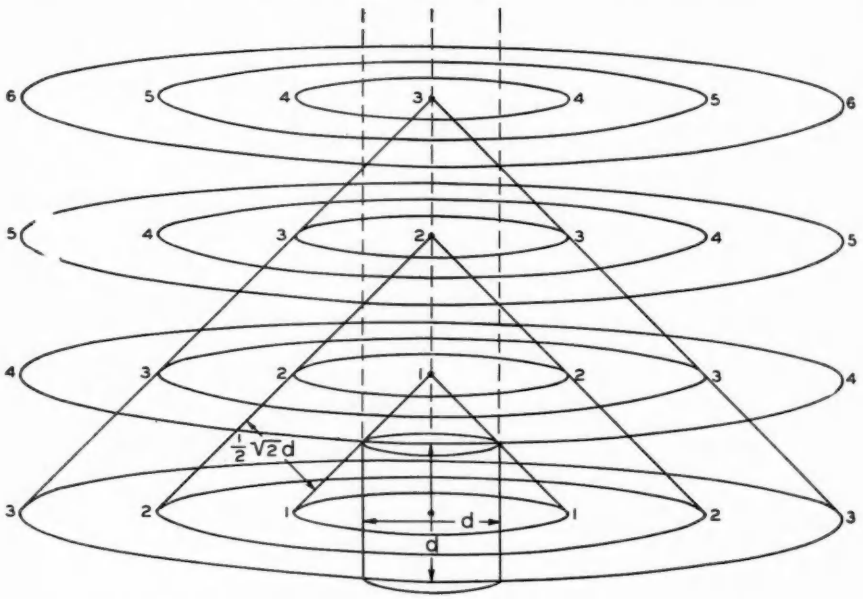

FIG. 2. *Illustrating a source at the center of a cylindrical room and a series of concentric circular line image sources.*

equation (5), the second one the amount given by the third term, etc., etc. When the source stops emitting energy all the image sheets do likewise. The persistence of sound is due to the fact that it takes time for the end of a sound wave which marks the termination of energy emission from a particular image sheet to reach the enclosure. The decay goes down in steps; due to the definition of reverberation time the end of the wave from a sheet located at a distance cT from the source passes thru the enclosure at the instant the intensity reaches one millionth of its original value.

Next consider a cylindrical room with diameter d equal to length, with uniform absorption over the surface and with a source at its center. Using the method of images we note that the source may be imagined imaged in the cylindrical walls as a series of concentric circular

line sources, and in the plane ends as a series of point sources, which in turn are imaged in the projected cylinder as concentric line image sources. (See Figure 2.) The numbers indicate the order of the reflection and if images of the same order are included in a surface, we shall have a series of concentric cones separated by a distance $\frac{1}{2}\sqrt{2}d$. If we identify this as the average distance between reflections[6] we may replace D

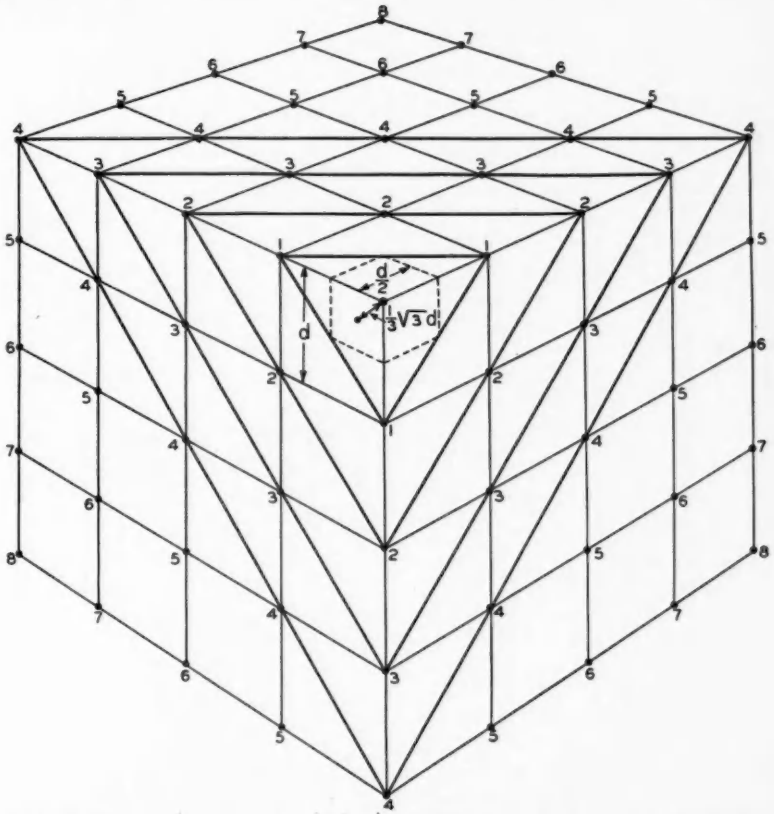

FIG. 3. *Illustrating a source at the center of a cubical room and a series of point image sources.*

of equation (6) by this value, and remembering that $d=6V/S$ for an enclosure of this sort we rewrite (6) and obtain the following decay equation for a point near the center of this cylindrical room,

$$S_\infty - S_n = \frac{3\sqrt{2E}}{S\alpha_a c}\, e^{\frac{cS \log_e (1-\alpha_a)t}{3\sqrt{2}V}}. \qquad (7)$$

The same method may be applied to a cubical room with the source

[6] Schuster and Waetzman loc. cit. have obtained these values by a more rigorous method.

at its center. The images are point sources and are located discretely throughout space as shown in Figure 3. As before the numbers indicate the order of reflection, and if the image sources of the same order are included in a given surface we get a series of plane surfaces separated by the distance $\frac{1}{3}\sqrt{3}\,d$, where d is the length of the cubical room. Identifying this distance as the average distance between reflections[6] and remembering that for a cube $d=6V/S$, we have from equation (6) the

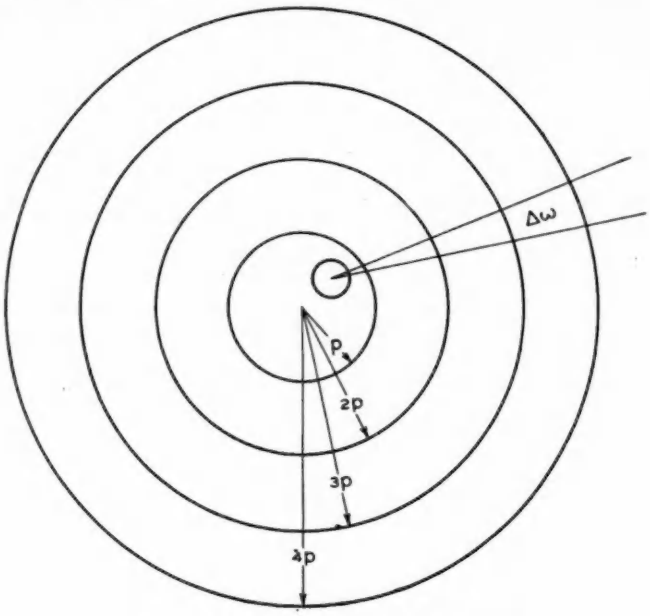

FIG. 4. *Illustrating zones of image sources.*

following decay equation for a point near the center of the cubical room,

$$S_\infty - S_n = \frac{2\sqrt{3E}}{S\alpha_a c}\, e^{\frac{cS\log_e(1-\alpha_a)t}{2\sqrt{3}\,V}}. \tag{8}$$

Formula (7) is valid for a cubical room with a source at its center provided two opposite walls are completely absorbing. The α_a in the coefficient of the exponential term is the average value for the whole room, but the α_a in the exponent of this term is the average value for the four reflecting walls. Again if the cubical room has two pairs of opposite walls completely absorbing equation (6) is valid, but just as before the proper values for α_a must be used.

So far we have considered rooms of very simple geometrical shape and

we have located the source at the center in each case. We now consider the more general case, make the shape more complex, move the source from the center and thus approach a diffuse condition of the waves. In the interest of simplicity we shall be content with average effects and assume that the image sources are discretely located in zones—surfaces of concentric spheres with radii, p, $2p$, $3p$, $\cdots np$, $\cdots$ etc., where p is the mean free path between reflections (See Figure 4). The mean free path is the average distance between reflections which the sound establishes on its many trips across the room from wall to wall, from wall to ceiling, from ceiling to floor, etc., as it travels in all conceivable directions. At present, the centers of these spheres are assumed located at the source of sound.

This means that we have replaced the source of sound and the walls of the room, by the source of sound surrounded by image sources located in evenly spaced discrete zones. To determine the energy density produced in the enclosure by any zone, we proceed as in the case of the spherical room and follow through the growth of sound in the room as follows.[7] The rate of energy emission is denoted by $\overline{E}$, c is the speed of sound, p/c is the average interval of time between reflections, and R is the coefficient of reflection already defined. The amount of energy emitted into the room directly from the source during this interval is $p\overline{E}/c$; the amount left over after the first reflection, or the amount emitted into the enclosure by the first zone of image sources during the same interval of time, is $p\overline{E}R/c$; and that emitted in the same time into the enclosure by the nth zone is $p\overline{E}R^n/c$. Because of the meaning of the mean free path p, it takes just this interval of time for the energy flow to fill the enclosure once it starts to enter it. At once the energy density in the enclosure due to the nth zone is

$$E_n=\frac{p\overline{E}}{cV}R^n. \tag{9}$$

It now becomes interesting to determine whether E_n is an average value or is constant for all parts of the enclosure. The sound intensity varies inversely with the square of the distance from the source or its images. The intensity of the direct sound, then simply obeys the inverse square law. To investigate the point to point variation within the enclosure of the sound energy density due to the various zones we proceed as follows. For simplicity neglect differences in phase and assume that

[7] Collected papers, Sabine. p. 43. Vibrating Systems and Sound, Crandall, p. 201.

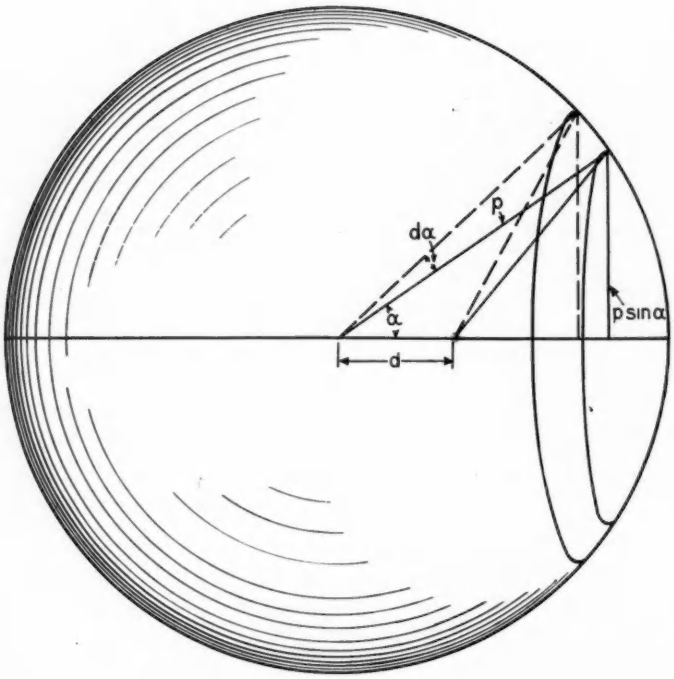

FIG. 5.

the image sources are evenly distributed over the zones. Call the surface density of image sources i. Then for the first zone, the radius of the sphere is p, and if we let d be the distance of the point of observation from the center of the sphere, then the energy density at the point of observation is given by the relation (See Figure 5),

$$E_1 = Mi \int_0^{\pi} \frac{2\pi p^2 \sin \alpha d\alpha}{p^2 - 2pd \cos \alpha + d^2} = \frac{Kp}{d} \log_e \frac{p+d}{p-d} \tag{10}$$

where M and K are constants.
Now expanding $\log_e (p+d)/(p-d)$ we get

$$E_1 = 2K\left[1 + \frac{1}{3}\left(\frac{d}{p}\right)^2 + \cdots\right]$$

and for any zone

$$E_n = 2K_n\left[1 + \frac{1}{3}\left(\frac{d}{n_i'}\right)^2 + \cdots\right]. \tag{11}$$

For all but the first zone, $(d/np)^2$ is small and E_n is essentially independent of d, the distance of the point of observation from the cen-

ter of the sphere. This means that neglecting interference any zone beyond the first, viz. the rth zone, contributes approximately the same energy to all points in the enclosure, and therefore, that the energy density produced by this zone is constant throughout the room, and of a magnitude E_r as given by equation (9). However, E_0 and E_1 are average values, the actual point to point variation of the energy density being obtained for each case by the use of the inverse square law and the method of images. The actual energy density at a point in the enclosure, then, may be divided into three parts: (1) that which is produced by the source only and this varies inversely as the square of the distance from the source, (2) that which is produced by the first reflection image sources, and the point to point variation of intensity may be calculated for each particular enclosure; (3) that which is produced by all the remaining image sources, a term with a magnitude $\sum_2^\infty E_n$, which shows no point to point variation.

In the following development of the sound intensity decay equation, we assume that the average densities E_0 and E_0R may be substituted for the actual densities produced at the point of observation by the source and the first reflection image sources respectively. Because of this the equation developed will not give an exact statement of the decay during the first two (p/c) sec. intervals, but this is not important because from then on it will give the correct statement and in good experimental practice the rate of decay obtained need never depend upon the changes during these first two intervals.

Hence when sound is established in the enclosure, the average density due to the source and all the zones of image sources is $S_\infty = \sum_0^\infty E_n$. Remembering equation (9), we have $E_0 = p\bar{E}/cV$ and at once

$$S_\infty = E_0(1+R+R^2+R^3+\cdots R^n+\cdots). \tag{12}$$

When the source is stopped all the image sources stop, and, of course, the energy supply also. At the source, or very near it, the portion E_0 of the total energy at once vanishes. The intensity then remains constant for an interval p/c seconds, the time it takes sound to travel from the first zone to the point of observation announcing the stopping of the energy supply of this zone. At the close of this interval the portion E_0R abruptly vanishes. This is followed by another interval of constant intensity. Then a sudden disappearance of the portion E_0R^2 occurs. By this time the sound has traveled from the second zone to the point of observation and announces the stopping of the energy supply of that

zone. Thus the decay continues and remembering that equation (12) is a geometric series, the energy density during the nth interval is

$$S_\infty - S_n = \frac{E_0}{1-R} - \frac{E_0(1-R^n)}{1-R} = \frac{E_0 R^n}{1-R} . \tag{13}$$

We developed this equation on the assumption that the point of observation was very near the source, which was also the center of the zones. There seems to be no *a priori* reason why the center should be chosen at the source rather than at some other point in the enclosure, but once the center is located it does not seem legitimate to move it about without a careful investigation of what would happen if it were not moved to each point of observation.

Hence we select for study some point of observation not at the center of the zones. Waves leaving a given zone at the same instant will not arrive simultaneously at this new location. This means that unless one judges the energy density at a point by the flow of energy out in a *given direction*, equations (12) and (13) will not give a correct picture. But the decay history at a point as witnessed in a *given direction* is given exactly by equation (13). We propose to determine the decay history of the energy density by *adding up the complete decay histories of all the directions*; we shall not obtain the decay history by adding the instantaneous values of the directional histories, and then tracing the history of this instantaneous sum. Therefore, let e_0, e_1, e_2, etc. be the energy density due to the image sources, included within any solid angle $\Delta\omega$. (See Figure 4). Then

$$e_0 = \frac{\Delta\omega}{4\pi} E_0, \quad e_1 = \frac{\Delta\omega}{4\pi} E_1, \text{ etc.}$$

By equation (13)

$$s_\infty = \sum_0^\infty e_n = \frac{e_0}{1-R}$$

$$s_n = \sum_0^n e_n = \frac{e_0(1-R^n)}{1-R}$$

and

$$s_\infty - s_n = \frac{e_0 R^n}{1-R} = \frac{\Delta\omega}{4\pi} \frac{E_0 R^n}{1-R} . \tag{14}$$

Now "n" is an integer and simply denotes during which p/c sec. interval the intensity is measured. Except in extreme cases, waves

which left simultaneously from a given zone will all arrive within the interval p/c sec. after the first arrival takes place. Hence we are justified in adding up equations (14) for all directions keeping "n" constant. At once

$$\sum(s_\infty - s_n) = \frac{E_0 R^n}{4\pi} \frac{\sum \Delta\omega}{(1-R)} = \frac{E_0 R^n}{1-R} \cdot \tag{15}$$

That this summation is legitimate, even for the extreme cases where a slightly greater interval than p/c sec. is needed to have all the waves arrive, is illustrated by Fig. 6, in which it is clear that the staggering effect simply smooths out the distinct drops of the *total* energy but does not change the slope of the decay. From this it becomes clear that so far as the energy density at a point is concerned, "n" which was introduced

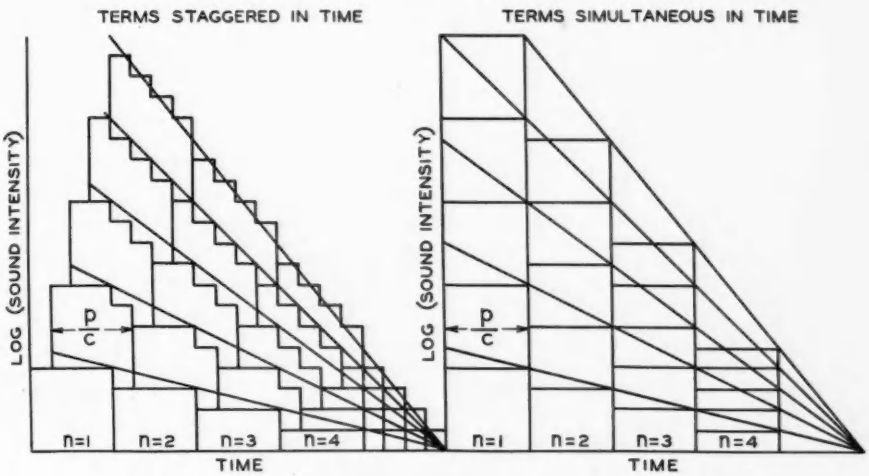

FIG. 6. *Illustrating that the staggering effect simply smooths out the distinct drops but does not change the slope.*

as an integer may after the summation is over, be considered, approximately at least, a continuous function. This method of tracing the "life history" (equation 14) for each particular direction and then adding the "life histories" is essentially the uniqueness of the new analysis.

In the old theory during the decay the energy density is assumed strictly continuous, but we have found this to be approximately true. The difference does not lie here. The old theory also assumes that the energy density at a point appears to have a continuous change as viewed from an element of surface. It tacitly assumes that an absorbing surface has the ability to register back into the oncoming wave the decrease in

intensity which it is causing. This is obviously not true; this particular absorption is registered in the outgoing wave, not in the oncoming wave. During decay the surface will receive *from a given direction* a constant energy flow for a time equal to the time it takes sound to travel the mean free path between reflections; then there will be an abrupt change. *This constant energy flow followed by an abrupt drop, rather than a continuous drop to this same level, means a greater absorption during the same interval of time and hence a more rapid decay of the sound.* (See Figures 7 and 8). This is in essence the physical difference between the foundation on which Sabine's formula has been erected, and the foundation on which we propose to erect a more general type of formula. The new type of equation must give Sabine's formula as a special case for "live" rooms and must do so simply because under these circumstances the two foundations become equivalent but not because the new picture ceases to hold for "live" rooms.

Rewriting equation (15) and remembering that

$$n=\frac{ct}{p} \tag{16}$$

we have

$$\sum(s_\infty - s_n)=\frac{E_0 R^{ct/p}}{1-R}$$

and remembering that

$$E_0=\frac{p\overline{E}}{cV}$$

we have

$$\sum(s_\infty - s_n)=\frac{p\overline{E}}{cV(1-R)}\, e^{\frac{c(\log_e R)t}{p}} \tag{17}$$

and from what we have said above we are justified in assuming that t is a continuous function.

We now have to determine p, the mean free path between reflections. Sabine[1] obtained the experimental value, $p=0.62(V)^{1/3}$, which reduces to $p=3.7V/S$ for enclosures of reasonably compact proportions. Jaeger[3] using the theory of probability and assuming a perfect diffuse condition of the sound waves calculated the number of reflections per second in the same manner that the impacts of molecules are estimated

in the kinetic theory of gases, and found $p=4V/S$. Recently Schuster and Waetzmann[5] have shown as we have also shown in the first part of this paper that for rooms of special shapes where an ordered not a diffuse condition of the sound waves is attained the mean free paths are as follows: cubical room, $p=2\sqrt{3}V/S$; cylindrical room with length equal diameter, $p=3\sqrt{2}V/S$; spherical room, $p=6V/S$. Probably the formula which will be used most is one based on a perfect diffuse condition of the waves, a special formula being written down for each enclosure which cannot fulfill this condition. To obtain the formula for the diffuse condition we shall use the value obtained by Jaeger for the mean free path. That this value may be used for "dead" rooms follows from the fact that Jaeger's analysis does not in any way involve the absorption coefficients of the surfaces. Of course, we must demand that the "dead" room shall produce a diffuse condition among the sound waves; the assumption that the image sources are located on the surfaces of concentric spheres tacitly implies this diffuse state.

Substituting the value

$$p=\frac{4V}{S} \tag{18}$$

in equation (17) we get the decay equation

$$\rho=\rho_0 e^{\frac{cS\log_e(1-\alpha_a)t}{4V}} \tag{19}$$

where

$$\rho_0=\frac{4\overline{E}}{cS\alpha_a}. \tag{20}$$

As we have already pointed out, for practical and experimental purposes, t may be considered continuous, and hence (19) may be thought of as derivable from a differential equation.[8] In a *very "live" room* (and in this analysis we may make the room as "live" as we please) the absorption per reflection is so very small that the energy density as viewed from an element of surface can be considered continuous. This means that we may follow the old theory[4] and write down the differential equation for the sound intensity in a closed room as follows:

$$V\frac{\partial\rho}{\partial t}+\frac{1}{4}c\alpha_a S\rho=\overline{E} \tag{21}$$

[8] Schuster and Waelzmann, loc. cit., have made use of a differential equation involving velocity potentials.

where V is the volume of the room, S its surface, α_a the average coefficient of absorption, c the speed of sound in air, ρ the energy density and $\overline{E}$ the rate of sound energy emission, which is considered constant.[9]

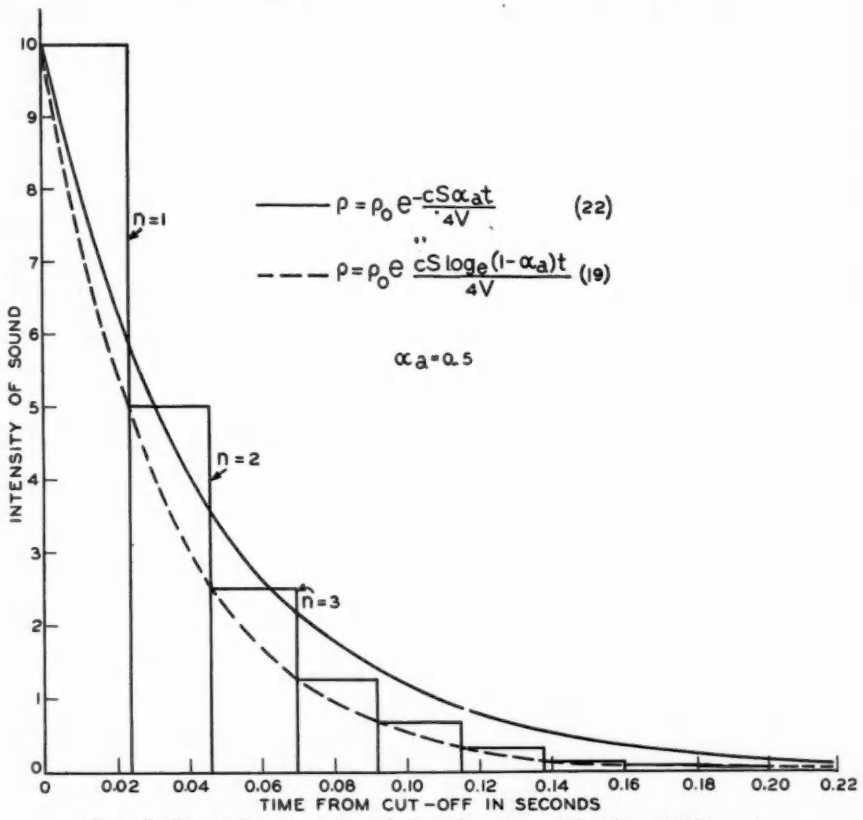

FIG. 7. *Illustrating rate of sound decay in a room with volume 80400 cu. ft., surface 12180 sq. ft. and walls of average coefficient of absorption 0.5.*

The solution of this equation under proper boundary conditions gives the following for the equation of decay

$$\rho = \rho_0 e^{-\frac{c\alpha_a S}{4V}t} \tag{22}$$

and from this Sabine's reverberation time equation is obtained. Now

$$\log_e R = \log_e (1-\alpha_a) = -\left[\alpha_a + \frac{\alpha_a^2}{2} + \frac{\alpha_a^3}{3} + \cdots\right] \tag{23}$$

and when α_a is very small as is true for *very live* rooms,

$$\log_e (1-\alpha_a) = -\alpha_a.$$

[9] Vibrating Systems and Sound, Crandall, p. 207.

This means that (22) is a special case of (19). The value of ρ_0 is the same for both equations and may be derived from differential equation (21) as the maximum value reached by the energy density as a steady state is established. This agreement is natural since the steady state value depends simply on the conservation of energy principle, and not upon the rate of growth or decay when the sound source is turned on or cut off. This differential equation cannot give the correct damping factor,

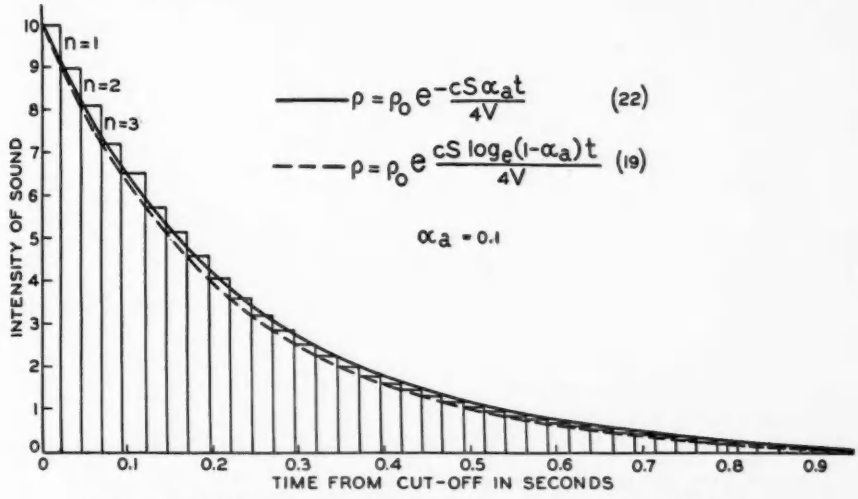

FIG. 8. *Illustrating rate of sound decay in a room with volume 80400 cu. ft., surface 12180 sq. ft. and walls of average coefficient of absorption 0.1.*

because as we have already pointed out, it does not describe the true nature of the absorption. However, if we put $\overline{E}=0$, substitute $-\log_e(1-\alpha_a)$ for α_a and set down the boundary condition, $\rho=\rho_0$ when $t=0$, we get a differential equation which gives (19) as its solution.

In Figures 7 and 8, we have plotted equation (19) first with t considered continuous, then with $n=ct/p$ considered as an integer. The decay is shown to be greater than that for equation (22), the old decay formula, the difference becoming very marked for "dead" rooms.

To obtain the reverberation time formula involving the true nature of the absorption we make use of equation (19) and write

$$\frac{\rho_1}{\rho_2}=\frac{e^{\frac{cS\log_e(1-\alpha_a)t_1}{4V}}}{e^{\frac{cS\log_e(1-\alpha_a)t_2}{4V}}}=10^6$$

and put $(t_2-t_1)=T$, the reverberation time.
At once

$$\log_e 10^6=\frac{-cS\log_e(1-\alpha_a)}{4V}T$$

$$T=\frac{-4\log_e 10^6 V}{cS\log_e(1-\alpha_a)}=\frac{0.05V}{-S\log_e(1-\alpha_a)}. \tag{24}$$

From equation (23) it follows at once that for very small values of α_a the reverberation time formula (24) becomes

$$T=\frac{0.05V}{S\alpha_a}=\frac{0.05V}{a} \tag{25}$$

which is Sabine's formula—*a special case of the more general formula* which we have developed.

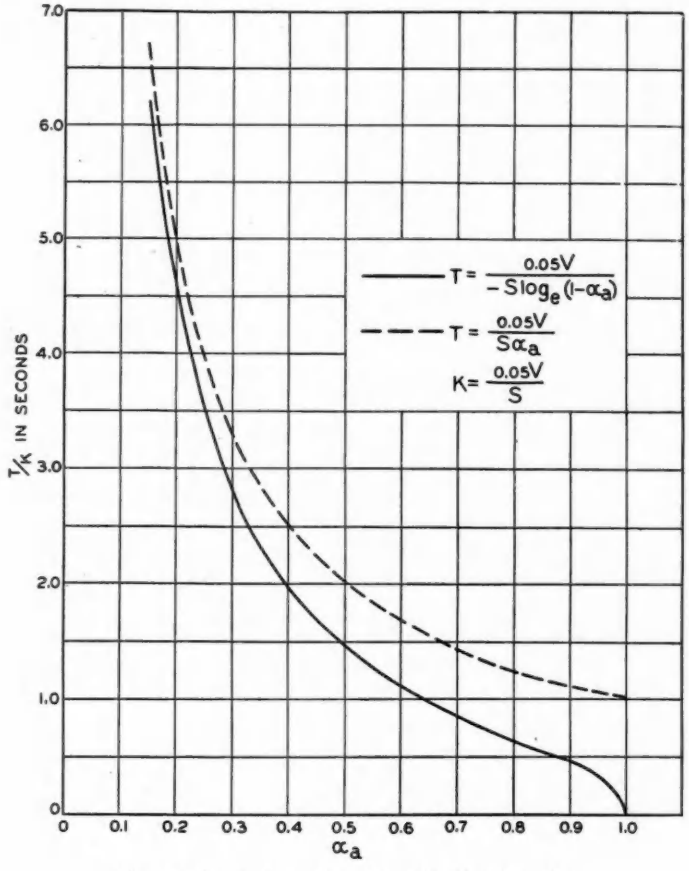

FIG. 9. *Comparison of reverberation time formulae.*

Although Sabine's formula had no meaning for $\alpha_a = 1$, complete absorption, the "new" formula does give the value zero, Curves comparing the new and old formulae are given as Fig. 9. These curves show a wide difference for large values of absorption but approach each other for small values of absorption. Thus we have established (24) as the general reverberation time equation for all types of rooms in which we can be certain that a diffuse condition of energy density is achieved. If we wish more generality, and certainly there will be cases as we have already pointed out where an ordered condition may exist among the sound waves, we shall not need to change the equation's form which is

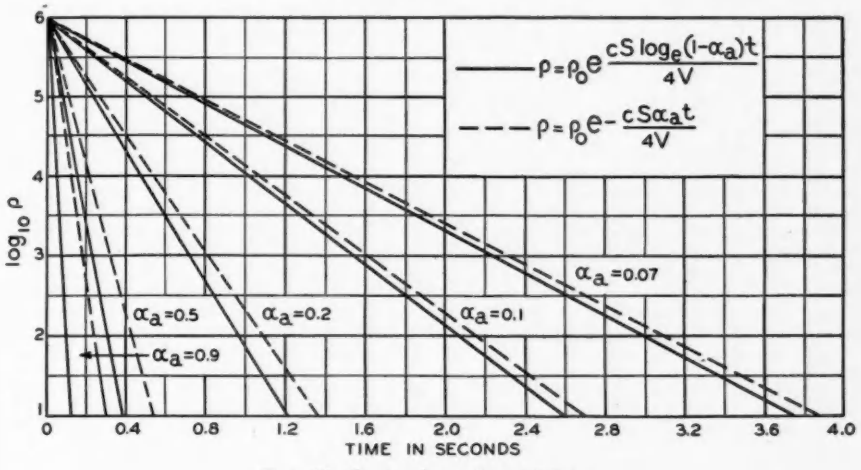

FIG. 10. *Comparison of sound decay.*

intimately tied up with the nature of the absorption, but we will need to substitute a constant k for 0.05 and then ascertain for each special problem the value of k to be used, remembering that the mean free path between reflections is the physical quantity that controls its magnitude. How the averaging represented by equation (3) is modified to obtain the proper value of α_a will depend on the nature of the ordered condition which exists among the sound waves. The method of images as used and discussed especially in the first part of this paper may be an aid in determining these two constants as may also the following table.

	Mean free path	*Constant k*
Sabine's experimental value	3.7 *V/S*	0.045
Statistical value	4.0	0.049
Cubical Room	3.5	0.043
Cylindrical room (length = dia.)	4.2	0.051
Spherical room	6.0	0.073

EXPERIMENTAL RESULTS

The fact that the experimental results obtained by Sabine checked his formula does not detract from the validity of formula (24) since the two formulae become identical for very live rooms, the type in which Sabine worked. We now wish to present evidence for the applicability of the general formula to work with very "dead" rooms and the failure

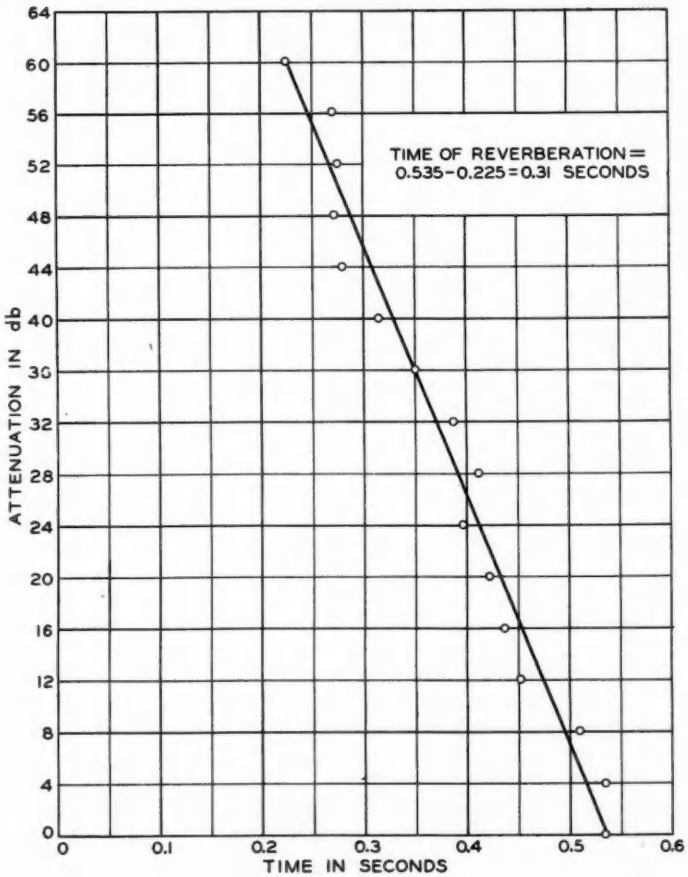

FIG. 11. *A typical experimental curve obtained in reverberation time measurements.*

of Sabine's formula for such cases. Formula (24) is used without modification because the shape of the room, and the location of the absorbing material, source and point of observation guarantee a diffuse condition of the sound waves. Equation (24) may be written thus

$$\log \rho = \frac{cS \log (1-\alpha_a)t}{4V} + \log \rho_0 \tag{26}$$

which shows that the time plotted against the logarithm of the sound energy density is a straight line. (See Figure 10). Since in a "dead" room the energy steps are rather marked, a picture of a decay will not be a smooth curve. The irregularity will also be greatly enhanced by interference phenomena which have not been considered in this paper. (See Figure 11).

Making use of equation (26) we may write

$$\log \frac{\rho_1}{\rho_2} = -K(t_2 - t_1). \tag{27}$$

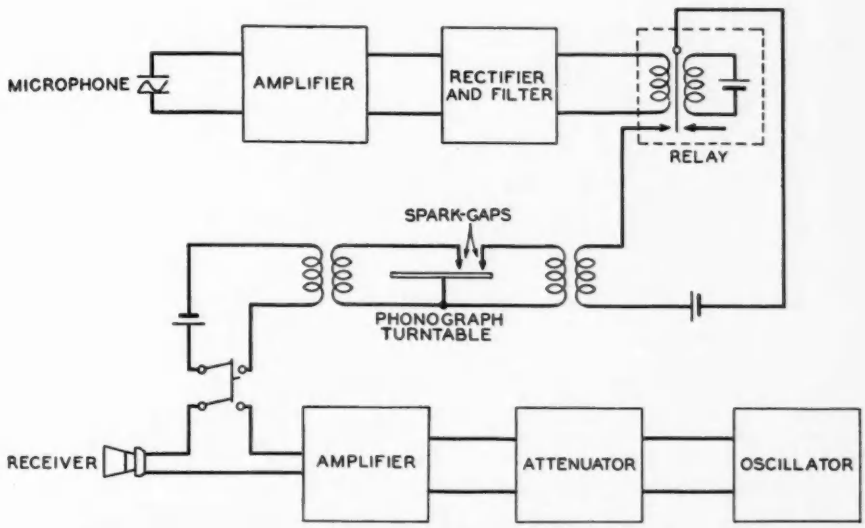

FIG. 12. *A block schematic of the reverberation time measuring apparatus used.*

If ρ_1 is the intensity of the sound at some specified instant, $t_1 = 0$, such as the time of sound cut-off, and if ρ_2 is a minimum intensity which marks a relay cut-off, then associated with each value of ρ_1 will be a certain time t_2. If, therefore, we give ρ_1 various values by changing the output of a receiver and obtain the corresponding values of t_2, and if the power input measured in *db* corresponding to each receiver output is plotted against each corresponding time between sound cut-off and relay action (t_2), we shall get a straight line with the same slope as equation (26). From this line the time for a 60 *db* drop may be obtained and this is the reverberation time. A typical curve obtained for the sound stage is shown in Figure 11. A block schematic of the apparatus is shown in Figure 12 and the plot of *db* attenuation of the maximum input is used instead of

the power input measured in *db*. The slope of the line, which is all we wish, is not modified by this procedure. The details of the apparatus and the technique of its use have been described by Wente and Bedell in the paper, "Chronographic Method of Measuring Reverberation Time."

TABLE I

INSULATING MATERIAL ONLY

Volume of Room 73,475 cu. ft. Total surface 11,553 sq. ft.
Ins. Material = 8401 sq. ft. Concrete, glass, etc. = 3,152 sq. ft.

Frequency	Reverberation Time	α_a	α for Insulating Material	
			New Formula	Sabine's Formula
120	0.73 sec.	0.35	0.48	0.59
240	0.49	0.47	0.65	0.89
500	0.38	0.56	0.77	1.14
1000	0.34	0.60	0.82	1.28
2000	0.34	0.60	0.82	1.28
4000	0.32	0.63	0.86	1.37

TABLE II

INSULATING MATERIAL AND MONK'S CLOTH CURTAINS

Curtain area = 1,670 sq. ft. Ins. material only 6887 sq. ft.
Concrete, glass, etc. 2,996 sq. ft.

Frequency	Reverberation Time	α_a	α for Combination
120	0.52	0.45	1.10
240	0.40	0.54	1.03
500	0.32	0.62	1.00
1000	0.31	0.64	1.01
2000	0.28	0.67	1.13
4000	0.26	0.70	1.27[1]

[1] No significance should be attributed to these values which are greater than unity since the additional absorption was introduced into a room which was already very "dead," and hence experimental errors are bound to be large.

We wish to emphasize the importance of making the time from sound cut-off to relay action sufficiently long so that enough reflections are included to give a good value of the mean free path between reflections. This means that for very high damping a time longer than the reverberation time will need to elapse between the stopping of the sound and the operation of a relay in the instrumental method of measuring reverberation time in order that "*n*" may be sufficiently large to insure the

use of a good number of terms of the geometric series. Only in this way will the walls have a chance to record their effect upon the sound decay. To make this more clear; suppose a source is 0.2 second from the point of observation and that the time of reverberation is 0.15 second. So long as the relay cuts off 60 *db* below the maximum intensity the only

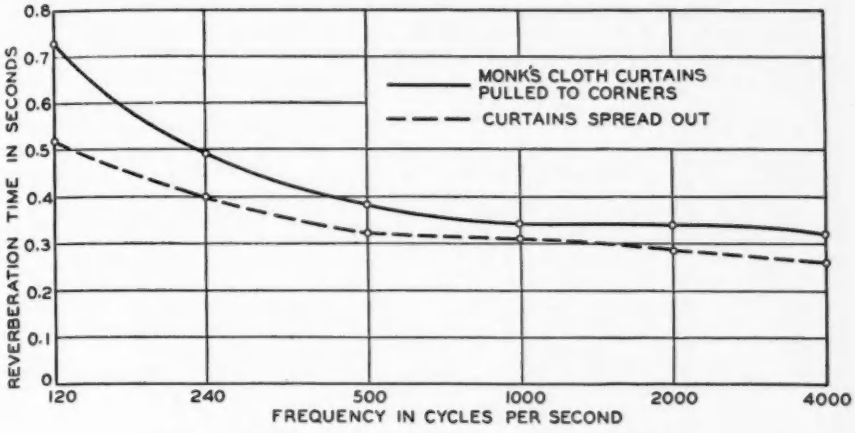

FIG. 13. *Reverberation time-frequency curves for sound stage.*

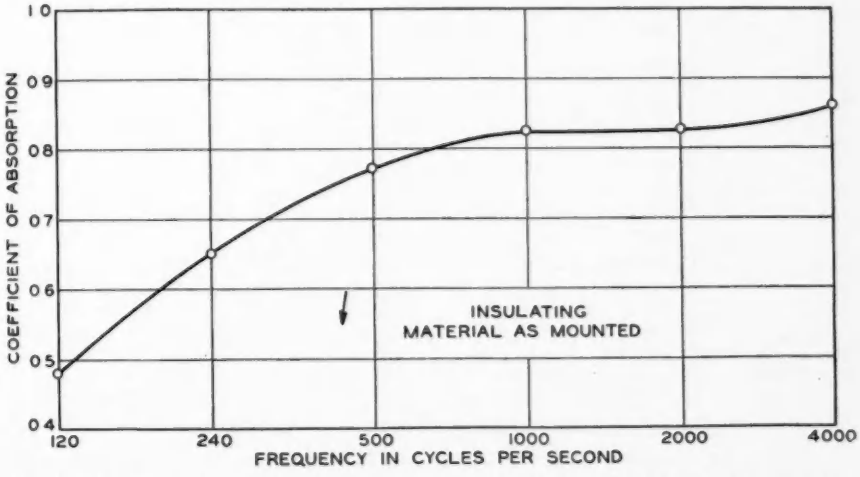

FIG. 14. *Coefficient of absorption-frequency curve for insulating material on walls and ceiling of sound stage, as calculated by the more general formula.*

time possible to measure is 0.2 second, the time it takes the sound to travel from the source to the point of observation. However, if the intensity of the sound is increased and if the threshold of the relay cut-off is lowered so that the time measured is for a 120 *db* drop, then the

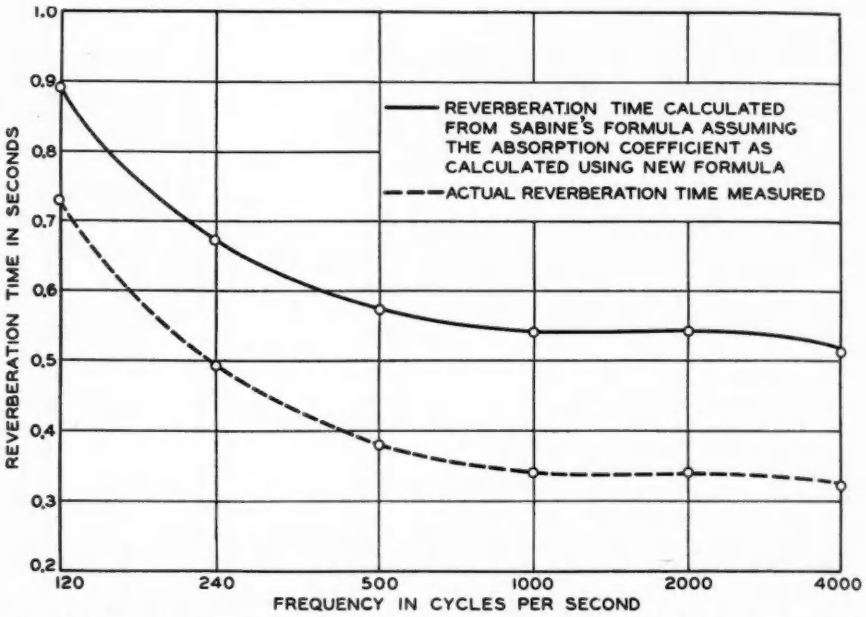

FIG. 15. *Comparison of reverberation times as measured and those calculated using Sabine's formula and coefficients of absorption shown in Figure 14.*

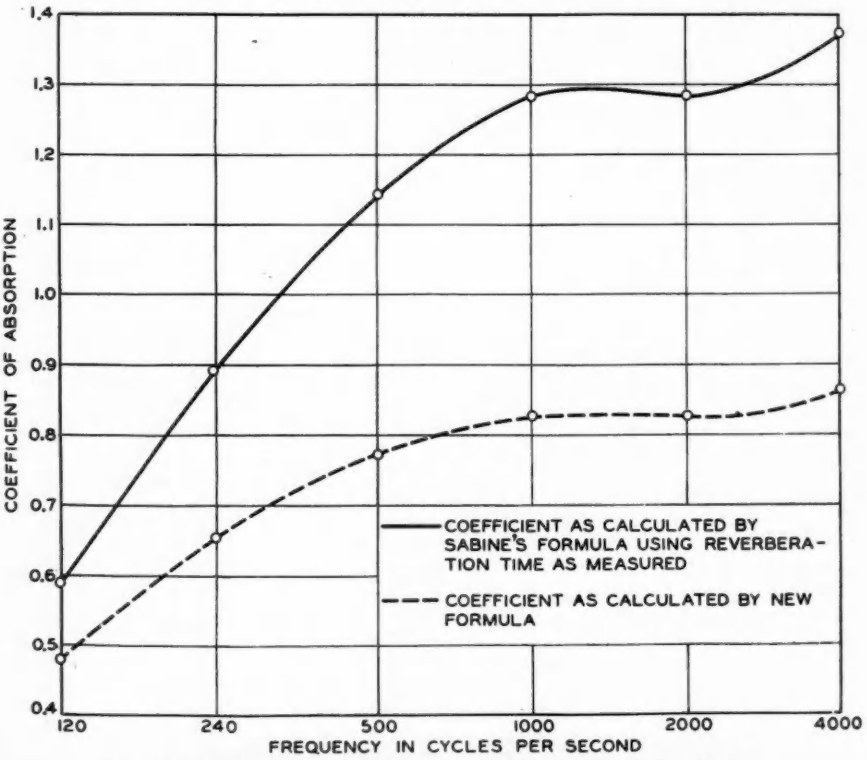

FIG. 16. *Comparison of coefficients of absorption as calculated by Sabine's and the more general formula using reverberation times as measured.*

time observed is 0.3 second and the reverberation time is 0.15 sec. Thus, a reverberation time approaching a zero value may be measured.

The results of measurements made in the Sound Stage, Sound Picture Laboratories, Bell Telephone Laboratories, Inc., are summarized in the preceding tables and illustrated in Figures 13 to 16.

Summary

We have shown that a correct picture of sound absorption in an enclosed room leads to a general reverberation formula of the form

$$T=\frac{kV}{-S\log_e(1-\alpha_a)}. \tag{28}$$

The manner of the absorption, which has been described in detail in this paper, is responsible for the logarithm factor and therefore the new form; the mean free path between reflections which is independent of the nature of the absorption controls the magnitude of k; and α_a is obtained by assigning proper weights to the various elements of surface. For a diffuse condition of sound energy, probably the condition most frequently encountered, $k=0.05$ and α_a is obtained from equation (3); but for each ordered condition of the waves a particular value of k must be used and α_a must be obtained by proper averaging. It is thought that the method of images presented in this paper will be of aid in solving each particular problem at hand. The practical importance of the new type formula should be emphasized because it shows that a "dead" room may be obtained by the use of less absorbing material than that calculated by Sabine's formula. An example will suffice. To obtain a required reverberation time of 0.54 second, an engineer using Sabine's formula would have specified 8400 square feet of absorbing material for a room such as the Sound Stage, but without knowing it, he would have obtained instead a reverberation time of 0.34 second; using the new formula he would have specified 6200 square feet of absorbing material securing a saving of 26% in material and the reverberation time desired.

OPTIMUM REVERBERATION TIME FOR AUDITORIUMS

By Walter A. MacNair
Bell Telephone Laboratories

I. Reverberation Time vs. Frequency

There is very little published data in regard to the change in reverberation time with frequency in auditoriums which are considered near ideal. It is often mentioned by engineers and physicists that to secure the best acoustical results, the reverberation time should be the same

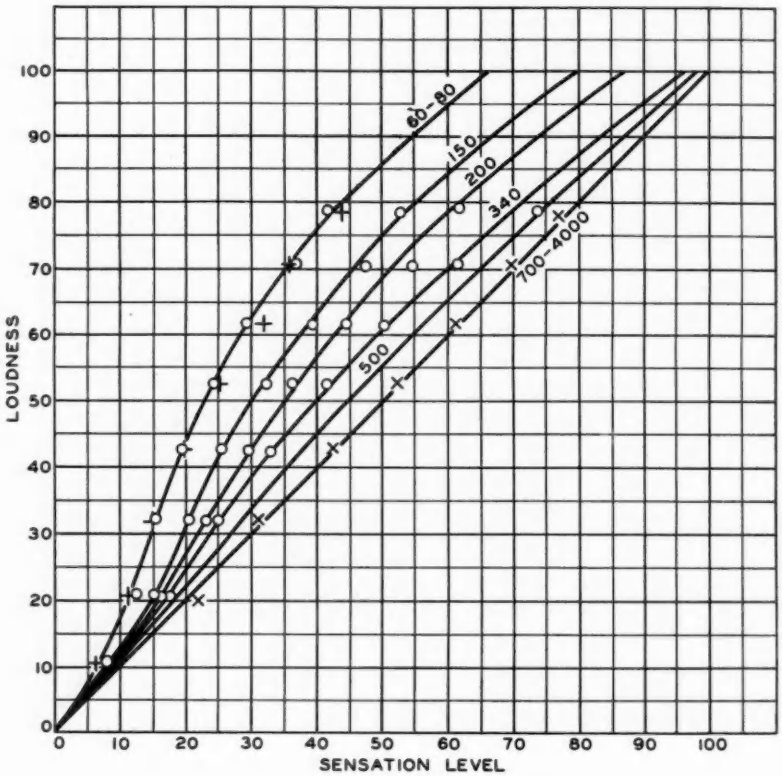

Fig. 1. *Loudness of Pure Tones.*

for all frequencies in any one room. This specifies that the sensation level shall decay at the same rate for all frequencies of interest.

It seems more reasonable, however, to specify that the loudness of all pure tones shall decay at the same rate for all frequencies since it is the loudness of a tone which takes into consideration not only the energy

level but also its ultimate effect upon one's brain. In Fig. 1[1] are plotted data which show the relation between the loudness as judged by a considerable number of observers and the sensation level. It will be seen that for frequencies between 700 and 4000 cycles per second these two quantities are equal to each other so that the two points of view mentioned above demand identical conditions throughout this frequency band. Outside of this band, however, any change in the sensation level gives a greater change in the loudness, as may be seen.

The maximum loudness in which we are interested at present is about 73.[2] In the figure the curves may be replaced by straight lines which represent fair approximations to the observed data up to this loudness. This family of straight lines may be represented by the expression

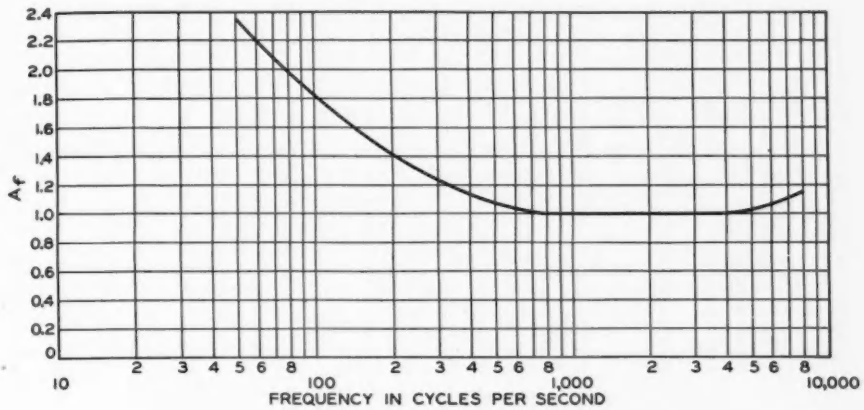

FIG. 2. *Value of A_f vs. Frequency.*

$$L_t = A_f S_t \qquad (1)$$

where A_f is the slope of the line adopted to fit the data for the frequency f. The values of A_f chosen from this figure are given by the next, Fig. 2. This approximation simplifies our calculations very much and introduces errors which are not intolerable.

Referring back to Fig. 1, if we wish to adjust the absorption of the room so that the loudness of all pure tones will decay at the same rate, say for the moment 60 units per second, it is seen that the sensation level must drop 60 *db* per second for frequencies between 700 and 4000 cycles and for other frequencies the sensation level must drop $60/A_f$ *db*

[1] This is Fig. 108 from "Speech and Hearing" by H. Fletcher.

[2] This is the loudness that the source chosen in Part II of this paper will produce in a room of 1000 cubic feet having a reverberation time of 0.8 seconds.

per second; or in other words, the reverberation time for frequencies between 700 and 4000 cycles should be one second and outside of this band it should be A_f seconds. Fig. 2, then, which is a plot of A_f vs. frequency now becomes also an illustration of the shape of the reverberation time vs. frequency curve which a room should have in order that the loudness of pure tones of all frequencies shall decay at the same rate.

According to Sabine's well known formula the reverberation time is inversely proportional to the number of absorption units in the room so

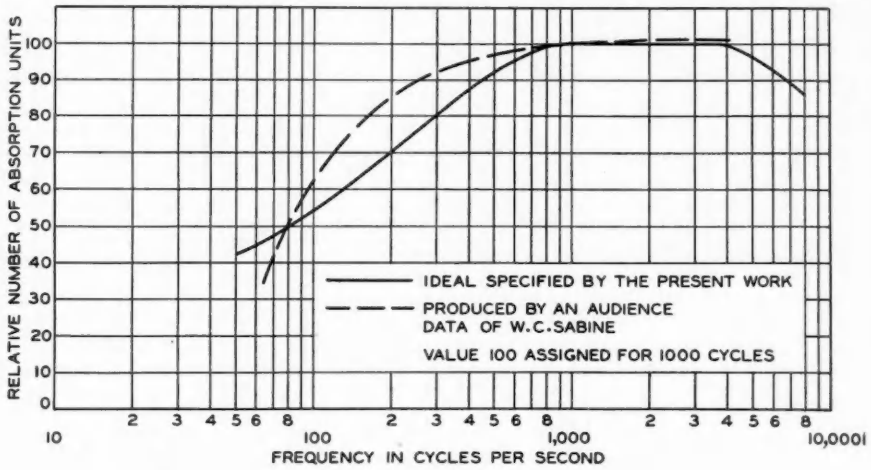

FIG. 3. *Relative Number of Absorption Units vs. Frequency.*

that, if we assume this, we may immediately infer the shape of the curve which represents the number of absorption units necessary at any frequency, referred to the amount required at 1000 cycles, to obtain our required condition. These values are plotted in Fig. 3. If it should happen that the greater part of the sound absorption in a room is caused by one particular kind of surface, then the curve in Fig. 3 is the shape of the absorption curve that this material should have.

A pertinent observation on which every one seems to agree is that if an auditorium has an unusually long reverberation time and consequently is of little use, when empty, it attains excellent acoustic conditions when filled with a large audience. In these cases a very large part of the absorption is caused by the audience. The absorption of an average audience has been measured by W. C. Sabine[3] and his results are also plotted in Fig. 3. The close agreement between this curve and

[3] "Collected Papers on Acoustics" page 86.

the one we have obtained from our hypothesis gives considerable confidence in our general viewpoint.

II. Reverberation Time vs. Volume

It is generally accepted that the best acoustical conditions in a room are obtained when the reverberation time is adjusted to a definite value known as the optimum reverberation time. Observations reported in literature agree that the value of the optimum reverberation time increases with the size of the room in the way shown in Fig. 4 where the curves numbered 1 to 3 are the choices reported by Watson,[4] Lifschitz,[5] and Sabine,[6] respectively. These experimental results have served as the basis of successful adjustment and design of many auditoriums. One naturally seeks the factor which determines a choice of reverberation time of two seconds for a million cubic feet theatre and on the other hand a choice of near one second for a 10,000 cubic foot music room. It is our purpose now to point out the factor which apparently does this.

We will set down a condition which we believe to be this factor and then will show that the requirements demanded by it agree quite closely with the empirical results illustrated in Fig. 4 and mentioned above. The condition is

$$\int_{t_0}^{t_1} L_t dt = -K \tag{2}$$

in which t_0 is the time a sustained source of sound $\overline{E}$ is cut off, t_1 the time the sound becomes inaudible, L_t the loudness of the sound at any instant t, and K a constant. As shown in Fig. 1, the loudness of a one thousand cycle note is equal to the sensation level, that is,

$$L_t = S_t \text{ for 1000 cycles.}$$

Since, during the time of decay, S_t decreases uniformly with time, and therefore L_t also, then for a thousand cycle note, evaluating our integral we have

$$L_{t_0} T_1 = 2K \tag{3}$$

or

$$S_{t_0} T_1 = 2K$$

where

$$T_1 = t_1 - t_0.$$

[4] Watson, Architecture, May, 1927.
[5] Lifschitz, Phys. Rev. 27, 618, 1926.
[6] Sabine, Trans. of S.M.P.E. XII 35, 1928.

This last expression is practically in the form in which this condition was first stated by Lifschitz.[5] In (3) there are three unknowns and a fourth is implied, namely, the power of the source, $\overline{E}$.

We now turn our attention to finding the relation between the volume of a room and the reverberation time dictated by the stated condition. Following P. E. Sabine let us take the rate of emission of the source, $\overline{E}$ to be 10^{10} cubic meters (35.3×10^{10} cubic feet) of sound of threshold density per second. Now[7]

$$T_1=\frac{4V}{ca}\log_e\frac{4\times35.3\times10^{10}}{c\cdot a}\cdot$$

Where V is the volume of the room in cubic feet.

c is the velocity of sound, 1120 feet per second.

a is the number of absorption units in sq. feet and[8]

$$L_{t_0}=S_{t_0}=10\log_{10}\frac{4\times35.3\times10^{10}}{c\cdot a}\cdot$$

If we should substitute these values in (3) we would obtain a relation between V, a, and K which must be satisfied when condition (2) is satisfied. In other words, (3) specifies the amount of absorption, for a one thousand cycle note, a room should have if it complies with (2). If we assume Sabine's well known formula, namely,

$$T=\frac{.05V}{a}$$

where T is the reverberation time in seconds we may express this relation in terms of V, T, and K with the result

$$10.40+\log T_{op}-\log V=\frac{(2K)^{1/2}}{1.283T_{op}^{1/2}}\cdot \qquad (4)$$

Where T_{op} is the value of T imposed by our condition (2) for a thousaud cycle tone.

Referring to Fig. 4 it will be seen that all three observers agree rather closely that the reverberation time for an auditorium of 1,000,000 cubic feet should be 2.0 seconds. This value refers to a tone of 512 cycles, the customary frequency used for experimental observation. It has been shown above that the reverberation time for 1000 cycles should be 92.5

[7] See Crandall "Theory of Vibrating Systems and Sounds" page 211.

[8] See Crandall "Theory of Vibrating Systems and Sounds" page 210, and the definition of sensation level.

per cent of the reverberation time for a 512 cycle tone, so that the 2.0 seconds above corresponds to 1.85 seconds for 1000 cycles. We can

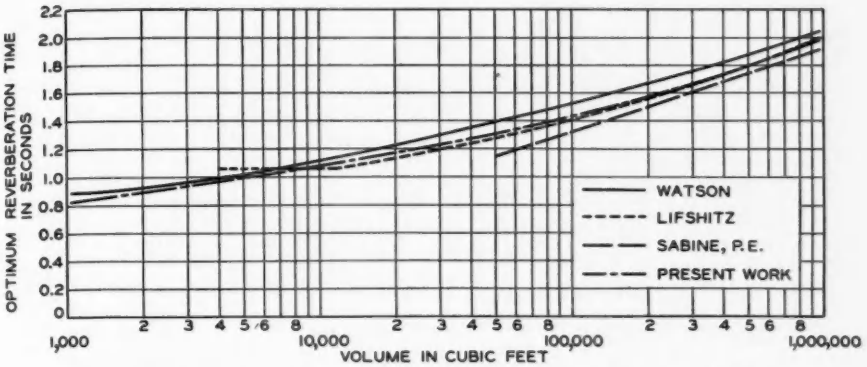

FIG. 4. *Optimum Reverberation Time vs. Volume in Cubic Feet for 512 Cycles.*

evaluate K in (4) by adapting this latter value of T_{op} for a volume of 1,000,000 cubic feet. This gives $K = 32.6$. Substituting this value in (4) we obtain

$$\log V = 10.40 + \log T_{op} = \frac{6.35}{T_{op}^{1/2}}. \tag{5}$$

From (5) we may obtain T_{op} for 1000 cycles for any volume. See Fig. 5. As mentioned above these values of T_{op} are 92.5% of T_{op} for 512 cycles so that these latter may be easily deduced for comparative purposes. These values are plotted to give curve number 4 in Fig. 4. It is seen that this curve agrees very well with those showing the choices of competent judges.

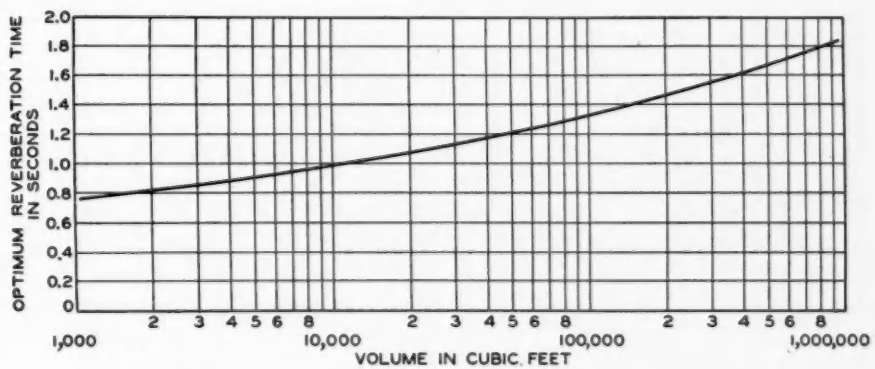

FIG. 5. *Optimum Reverberation Time vs. Volume in Cubic Feet for 1000 Cycles.*

III. The More General Hypothesis

Equation (2) may be written as follows, since we have assigned a value to K:

$$\int_{t_0}^{t_1} L_t dt = -32.6 \tag{6}$$

and it will be remembered that we have considered L_{t_0} to be the loudness set up by a certain standard source. Allowing V to vary with f constant (1000 cycles) we have obtained a relation between the optimum reverberation time and volume of rooms for 1000 cycles. We wish to point out now that exactly this same condition (6) with V constant and f variable, will give the same results that we have obtained in Part I of this paper with the only further requirement that for other frequencies than 1000 cycles the strength of the source $\overline{E}$ shall be such that the loudness L_{t_0} set up in the room at the frequency considered shall be exactly the same as the loudness which our standard source would set up at 1000 cycles.

In Part I of this paper our stated condition was that the loudness of all pure tones shall decay at the same rate for all frequencies. Since we have specified that the loudness at the time t_0 shall be the same for all test frequencies and also that the loudness at the time t_1 shall be zero for all frequencies, it is quite evident that the above integral can have the same value at all these frequencies only when the loudness decays at the same rate for all frequencies concerned. In other words, this condition stated as an integral specifies exactly the same requirement on the decay of loudness that we expressed in our statement early in Part I of this paper.

IV. Conclusions

To recapitulate, we have set down an equation, together with a specification of the strength of the virtual source in each case, from which we obtain the value the reverberation time for any frequency tone should have in any sized room according to the condition which apparently controls the choice of observers.

One naturally turns to see what meaning may be attached to this significant expression, namely, the integral of the loudness taken throughout the time of decay to inaudibility. Since this integral has the same values for all auditoriums which are considered ideal, it implies that one's brain is a ballistic instrument which is concerned with not only the maximum value of loudness but also with the effect of loudness integrated throughout a considerable interval of time.

A NEW JUST SCALE

With Proof that an Additional Just Scale is Impossible

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The just scale whose intervals in ascending order are 9/8, 10/9, 16/15, 9/8, 10/9, 9/8 and 16/15 was first described by Claudius Ptolemey in his "Harmonics" in the first half of the second century. The purpose of the present paper is to describe a second just scale, to compare its value with that of the Ptolemaic scale, and to show that no other seven tone scale of equal value is possible.

A scale may perhaps be defined as a division of the octave into intervals suitable for musical purposes. We also define the octave as the interval between the first and second partials of the harmonic series,

$$O=2/1,$$

the perfect fifth as the interval between the second and third partials of that series,

$$F=3/2,$$

the perfect fourth as the interval between the third and fourth partials,

$$f=4/3,$$

the major third as the interval between the fourth and fifth partials,

$$T=5/4,$$

the minor third as the interval between the fifth and sixth partials,

$$t=6/5,$$

the major sixth as the interval between the third and fifth partials,

$$SX=5/3,$$

and the minor sixth as the interval between the fifth and eighth partials,

$$sx=8/5.$$

All these intervals the ear regards as consonant, probably for the reason that, lying comparatively low in the harmonic series, the partials defining them are relatively strong and the ear has therefore become better adapted to them than to intervals defined by higher, weaker partials. If this surmise is correct, then we have an answer to

the ancient query of Pythagoras why notes produced by strings whose relative lengths are represented by small numbers are more pleasing to the ear when sounded together than notes produced by strings whose relative lengths must be represented by larger numbers.

But while the preceding intervals are consonant, the intervals of the major tone, lying between the eighth and ninth partials,

$$M=9/8,$$

the minor tone, between the ninth and tenth partials,

$$m=10/9,$$

and the semitone, between the fifteenth and sixteenth partials,

$$s=16/15,$$

the ear does not accept as consonant. It should be borne in mind that the term "interval," while convenient for discussion, has no concrete meaning except as expressing a ratio between the relative frequencies of the partials defining the "interval," so called; in consequence, the equations defining the above intervals express also the relative frequencies of their defining partials.

Now since

$$(9/8)(10/9)(16/15)(9/8)(10/9)(9/8)(16/15)=2,$$

therefore, substituting,

$$MmsMmMs=0 \tag{1}$$

or

$$M^3m^2s^2=0 \tag{2}$$

which latter is the fundamental equation of the seven tone scale employing the major tone, minor tone and semitone, and no other, as primary intervals. That equation (2) is fundamental to such a scale is easily shown: if p, q and r represent respectively the number of major tones, minor tones and semitones in the seven tone scale, then we have the indeterminate system,

$$p+q+r=7$$

and

$$(9/8)^p(10/9)^q(16/15)^r=2,$$

whose only integral solution is

$$p=3,\ q=2 \text{ and } r=2.$$

While it thus appears that major tones, minor tones and semitones can enter the seven tone scale only if three major tones, two minor tones and two semitones are employed, no limitation has yet been established regarding the order in which they shall be arranged, for, by the commutative law, the product is unaffected by the order of the factors. But of the many possible permutations of these seven factors a small number indeed fall within our definition of a scale as "a division of the octave suitable for musical purposes" because of its closing qualification, "suitable for musical purposes." For the past six centuries, at least, the Western world has regarded consonance as a qualification highly desirable in the intervals of a scale "suitable for musical purposes." From this consideration it follows that the number of consonant intervals within the octave should be as large as possible; and the application of this criterion rules out all but a few of the permutations possible to three major tones, two minor tones and two semitones. When any permutation is under consideration the products obtained by combining any number of succeeding intervals should give as many consonant and as few dissonant intervals as possible.

Now the major tone followed by the semitone combine to form a minor third,

$$Ms = (9/8)(16/15) = 6/5 = t,$$

a consonant interval. Likewise

$$Mm = (9/8)(10/9) = 5/4 = T,$$
$$Mms = (9/8)(10/9)(16/15) = 4/3 = f,$$
$$M^2ms = (9/8)^2(10/9)(16/15) = 3/2 = F,$$
$$M^2ms^2 = (9/8)^2(10/9)(16/15)^2 = 8/5 = sx,$$
$$M^2m^2s = (9/8)^2(10/9)^2(16/15) = 5/3 = SX,$$

all consonant intervals. The minor tone and the semitone combine into an interval which I have ventured to call the Doric third,

$$ms = (10/9)(16/15) = 32/27 = d,$$

and two major tones, two minor tones and two semitones, equal to two perfect fourths, combine into the minor seventh,

$$M^2m^2s^2 = (9/8)^2(10/9)^2(16/15)^2 = 16/9 = sv.$$

Both the Doric third and the minor seventh are sufficiently consonant, perhaps, to be admitted to that category.

On the other hand the intervals,

$$M^2m=45/32,$$
$$Mms^2=64/45 \text{ and}$$
$$M^3m^2s=15/8,$$

respectively the augmented fourth, the diminished fifth and the major seventh, together with such intervals as

$$M^2=81/64,$$
$$m^2=100/81,$$
$$s^2=256/225,$$
$$T^2=(Mm)^2=25/16 \text{ and}$$
$$t^2=(Ms)^2=36/25,$$

and many other combinations, give dissonant intervals which militate against the musical suitability of the scale under consideration, and indicate combinations to be avoided. For, since the number of intervals in a seven tone scale is constant, it is obvious that the introduction of any dissonant interval decreases the number of possible consonant intervals.

Now since any two of the three primary intervals combine into a consonant interval if the pair are unlike, e.g., $M\ m$, $M\ s$ and $m\ s$, and into a dissonant interval if the pair are alike, e.g., M^2, m^2 and s^2, therefore no major tone, minor tone or semitone should follow itself in the scale. Nor should a major third or a minor third follow itself, since T^2 and t^2 are likewise dissonant intervals. The perfect fourth may follow itself, however, since two perfect fourths combine into a minor seventh, a consonant interval. Indeed it is impossible to avoid repeating the fourth if the repetition of the major tone, minor tone and semitone is avoided: for since the fundamental equation,

$$M^3m^2s^2=0,$$

may be written in the form

$$M(Mms)^2=0$$

or

$$Mf^2=0,$$

it follows that the repetition of the fourth is inevitable no matter where the third major tone may be placed in the scale.

If, now, we arrange these seven intervals in all possible ways, with a major tone in the middle and a tetrachord on each side, avoiding the

placing of another major tone either immediately preceding or following the middle major tone, we have the following sixteen possible arrangements:

1)	$M\ m\ \underline{s}$	$\underline{M}$	$\underline{s}\ \underline{M}\ m$	
2)	$\underline{M}\ m\ s$	M	$s\ m\ \underline{M}$	
3)	$\overline{M}\ m\ s$	M	$m\ M\ s$	(1)
4)	$\underline{M}\ m\ s$	M	$m\ s\ \underline{M}$	
5)	$\underline{m}\ M\ s$	M	$s\ M\ \underline{m}$	
6)	$\underline{m}\ \underline{M}\ s$	M	$s\ \underline{m}\ \underline{M}$	
7)	$\overline{m}\ M\ s$	M	$m\ M\ s$	(3)
8)	$m\ M\ s$	$\overline{M}$	$m\ s\ M$	(1)
9)	$s\ M\ m$	M	$s\ \overline{M}\ m$	(1)
10)	$s\ M\ m$	M	$s\ \overline{m}\ M$	(3)
11)	$\underline{s}\ M\ m$	M	$m\ M\ \underline{s}$	
12)	$s\ \underline{M}\ \underline{m}$	$\underline{M}$	$\underline{m}\ s\ M$	
13)	$M\ s\ \overline{m}$	M	$s\ M\ m$	(3)
14)	$\underline{M}\ s\ m$	M	$s\ m\ \underline{M}$	
15)	$M\ s\ \underline{m}$	$\underline{M}$	$m\ \underline{M}\ s$	
16)	$\underline{M}\ s\ m$	M	$m\ s\ \underline{M}$	

The underscored letters indicate that the minor third, $s\ M$, is repeated in the first arrangement; the major tone is repeated in the second, fourth, fourteenth and sixteenth; the minor tone is repeated in the fifth; the major third is repeated in the sixth and twelfth; and the semitone is repeated in the eleventh. All these arrangements must be discarded as having fewer consonant intervals than the remaining arrangements which repeat no interval except the fourth. The only arrangements remaining are the third, eighth and ninth, each labelled with a (1), and the seventh, tenth and thirteenth, each labelled with a (3). The arrangements labelled (1) all consist of the same sequence of intervals but beginning at different points of the cycle, the beginning

point in each being the overscored letter; and the same is true of the arrangements labelled (3). The sequence

$$MmsMmMs \qquad (1)$$

is the Ptolemaic scale, and the sequence

$$mMsMmMs \qquad (3)$$

is the new scale.

It is now established that no other seven tone scale can equal these two in the number of consonant intervals in the octave provided major tones, minor tones and semitones, and these only, are used as primary intervals. But no other intervals than the major tone, minor tone and semitone can be used as primary intervals in a consonant seven tone scale: for the only consonant intervals less than an octave are the major, minor and Doric third, the perfect fourth and fifth, the major and minor sixth, and the minor seventh; and when the octave is divided *consonantly*, first into the fourth and fifth, then into two fourths and a major tone, then each fourth divided into major and minor thirds, to a point where the division has resulted in a scale of seven intervals, the result is inevitably three major tones, two minor tones and two semitones, no matter in what manner the division shall have proceeded. It follows, then, that no other seven tone scale can have an harmonic value equal to the scale of Ptolemey and to the new scale here presented. It remains to examine the new scale somewhat further, and to compare its musical value with that of the Ptolemaic scale.

It is apparent that the new scale differs from the Ptolemaic only in that the first two intervals are interchanged, making the two tetrachords alike in the new scale. In this respect the new scale is superior to the Ptolemaic; for when the scales are produced on instruments of indefinite intonation the two tetrachords of the Ptolemaic scale differ by amounts so small that musicians find it difficult to differentiate between them and, in consequence, neither tetrachord is produced with precise intonation and the resulting harmonies are inaccurate; while the tetrachords of the new scale, being alike, lend themselves readily to precise production and, thus, to purer harmonies.

Except for this melodic symmetry in the two tetrachords of the new scale and the resulting purer harmonies, the harmonies of the two systems possess equal attractiveness: each scale has three major thirds, three minor thirds and one Doric third, the Doric third of the Ptolemaic scale (*D* to *F*) becoming minor in the new, and the minor third on the

leading tone of the Ptolemaic scale (*B* to *D*) becoming Doric in the new; each scale has a dissonant triad on the leading tone (*B*); the Ptolemaic scale has three major and two minor triads, while the new scale has two major and three minor; the triad on the supertonic (*D*) becomes minor in the new scale with the ratios 10:12:15 instead of Doric minor, as in the Ptolemaic, with the ratios $10^1/_7$:12:15; on the other hand, the triad on the dominant (*G*), which was major in the Ptolemaic scale with the ratios 4:5:6, becomes what might be called Doric major in the new scale with the ratios $4:5:5\frac{2}{7}\frac{5}{7}$. The seventh and higher chords are, upon the whole, equally dissonant in each system.

Summarizing, the new scale, whose intervals in ascending order are *m M s M m M s*, is harmonically equal to the Ptolemaic scale in all respects and superior to it in that its symmetrical tetrachords are conducive to precise intonation and pure harmonies, while no other seven tone scale of harmonic value equal to these is possible.

THE EFFECTS OF NOISE:

A summary of experimental literature

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This survey of experimentation dealing with the effects of auditory stimuli upon living beings cites only work which reports data. Opinions and estimates—in which the field abounds—have not been considered as data.

The experiments to be summarized have been done under a wide variety of auspices. In very few instances were they designed primarily with reference to practical acoustical problems, the acoustical features being, in the main, by-products.

There is also a great lack of scientific control and specification of the acoustical stimuli which were used. Decibel levels and frequency patterns are rarely mentioned. In consequence, it is difficult to make immediately practical interpretations of office, factory, and city noise effects from these experiments.

The variety of work to be reported, however, indicates very definitely that noise, as commonly understood, has widespread effects upon the living organism. In general, these effects are not desirable. These experiments, which in most instances have not been directed primarily to acoustical problems, give encouragement to direct acoustical approaches to these effects.

1. Effects on Motor Functions

The time taken to move the finger in response to a signal was measured in thousandths of a second without the presence of auditory distractions and then under the distraction of sounds produced by a metronome alone, a bell alone, and a tuning fork. The distracting auditory stimuli lengthened the reaction time, the bell showing the greatest lengthening effect, while the sound of the tuning fork was easiest to become habituated to. The experimenters report further that "the effect of a distractor upon the sensory reaction is dependent upon the conscious attitude of the moment." (1)

Reaction time has also been found to be less when minor musical chords are present than for the major chords. (2)

Lombard's classical experiments upon the extent of the knee-jerk

showed a variation in amplitude of the jerk under different auditory stimuli. A swing of 74 mm. in a quiet room increased to 105 mm. when Beethoven's "Funeral March" was played on a piano, and when Chopin's "Raindrop Prelude" was played the swing was reduced to 68 mm. (3) These results are characteristic of most of the effects obtained from musical stimuli inasmuch as the effect seems to vary with the nature of the selection being played.

Measurements of the time of the knee-jerk show that it is quicker when noise stimuli are present. (4)

Speed of tapping with the hand has been found to be slightly increased after listening to jazz music compared with this speed after listening to classical music. The same investigators report an advantage of about 2% in strength of grip after listening to jazz music, but this advantage disappears after twenty-five periods of listening to the same classical and popular selections. (5, page 218)

Ability to exert voluntary pressure on a dynamometer has been found to vary with the pitch of tuning fork stimuli, as well as with the distance of the fork from the experimental subject. In general, the louder the auditory stimulus the greater the voluntary effort. (6)

Handwriting made under the influence of a tone of low pitch is increased in size, whereas writing made while a high pitch was sounded was smaller and the letters were more angular. (7)

2. Effects on Simple Cerebral Functions

Visual sensations were intensified by sounds from a tuning fork which was vibrating while observers were trying to discriminate slight changes in color. (8)

Sensations of smell have also been found to be increased in intensity by the presence of noises. (9)

Patches of color at such a distance from the observer that they cannot be recognized become perceptible when a tuning fork is sounded. (10)

The presence of music has also been found to increase the accuracy of discrimination of lifted weights. (11)

Accuracy in immediate memory for nonsense syllables was increased 15% and delayed memory increased 8% by reducing a complex noise from 50 decibels to 40 decibels. (12)

3. Effects on More Complex Cerebral Functions

Intelligence tests taken by 56 university students under conditions of relative quietness and later with the auditory stimuli from two electric bells being sounded intermittently gave equivocal results, but "apparently the better men were actually hindered by the ditractors and the poorest men were aided." (13)

Another intelligence test was given to nearly three hundred college students who were divided into two groups. Each group took alternate forms of the Army Alpha test twice. The control group took it under relative quiet each time. The experimental group took it first under relative quiet and later under extremely noisy conditions. The control group had an improvement in score of 10.1 the second testing; the experimental group had an improvement the second testing of 6.4 points. (14) "If the expermental group had made a gain of more than 10.1 points there would have been evidence of increase in efficiency . . . . if the gain of the experimental group had been less (as was actually the case) the evidence would be that the noise actually played a detrimental role, although it would not eliminate the practice effect." (15)

A reduction of a complex noise from 50 decibels to 40 decibels resulted in an increase of slightly more than 30% in speed of mental multiplication of three-place by three-place numbers. No changes in accuracy were noted. (12)

A learning test in which ten keys of the typewriter were used to translate a code showed no difference in this type of learning when the room was noisy. Recorders attached to the keys, however, indicated that more muscular pressure was exerted on the keys during the noisy phases of the experiment. (16, 17)

4. Effects on Respiratory Functions

Breathing of frogs is slowed by relatively pure acoustic stimuli when the tones are prolonged, while the repeated stimuli of an alarm clock bell yield a faster breathing rate. Frogs with the cerebral hemispheres removed give the same results. With mammaliam forms of life (cat, dog, and rabbit) all acoustic stimuli produced faster and shallower breathing. Animals with the cerebral hemispheres removed gave the same results as those with intact nervous systems. (18)

Human beings in response to the unexpected sounding of an electric automobile horn breathe faster, after a momentary checking of the rate. The breathing is also deeper and the rate is irregular. (19)

5. Effects on Cardiovascular Functions

The noise of a firecracker raised systolic blood pressure 20 mm. in 20 seconds. (20) The same experimenter reports that music, both popular and classical, decreased the pressure. Another experimenter records that music speeds or slows the pulse, depending upon the character of the music. (21) Another reports that the noise made by a telephone bell will accelerate the heart rate. (22) Listening to popular music gave a pulse rate of 2.5 more beats per minute than listening to classical music. (5, page 218) The sound of a nearby electric automobile horn has been found to slow the pulse rate. (19) The pulse rate of dogs is increased as a result of auditory stimulation, even when the animal is under the influence of a strychnine. (23)

Increases in cardiac tension and acceleration of the pulse from 10% to 100% have been observed in dogs as a result of such noises as slamming a door, striking a stove with a shovel. While the effects were evident in practically every observation, they did not run consistently for each animal or from animal to animal. (24)

Due to an accident which made the brain membranes visible one observer was able to study the brain circulation under the influence of various conditions. It was found that the brain blood pulsations were increased after a musical note, and that high notes produced the greatest change. Actual musical compositions gave rather complicated and inconsistent results. (25)

6. Feeling-tone and Emotional Functions

A loud noise or bang is reported by trained introspective observers as causing unsteadiness and visual images of falling objects or explosives. (26) Visible fear reactions are readily observed in children in response to a loud noise; this reaction is shown by "a sudden catching of the breath, clutching randomly with the hands, sudden closing of the eye-lids." (27, page 200)

To musical stimuli it is reported that the feeling-tone "of the tonal stimulus is, other things being equal, independent of the individual." This same experiment also reports a tendency for mankind to find musical stimuli most pleasant when they are moderate in pitch, moderate in intensity, and moderate in duration. (28)

The mood effects of musical compositions have been shown to be marked, even in the case of unmusical listeners. (5)

It is reported that relaxation is difficult, if not impossible, in a noisy environment. (29)

The annoyance produced by stimuli from the #2 A audiometer has been found to increase fairly directly with increase in loudness, except that for the higher intensities annoyance becomes in part a function of pitch as well. (30)

7. EFFECTS ON THE ORGANISM AT WORK

Practical acoustical workers are probably more interested in effects of noise upon the organism at work than upon the basic psychological and physiological details which have been presented to this point. The preceding material has been of importance, however, in showing the fundamental effects on reactions and functions which are involved in the work of the organism as a complex entity.

A group of office workers engaged at a variety of machine operations in the home office of an insurance company showed a 12% increase in output when the noise level was decreased from 45 decibels to 35 decibels. No other changes were made in the office. The workers were on a bonus system for which production records were kept regularly. This increase in output is so great that the officials are inclined to attribute some of it to added skill from practice, although the workers were experienced at the beginning of the year of experimental observation. (31)

Experienced typists increased their output 4.3% when the noise level of their work room was reduced from 50 decibels to 40 decibels. There was no consistent tendency with reference to errors. The typists slowed up in the noisy room during the two hour test periods, and had a tendency to warm up and gain speed in the same room 10 decibels quieter. (32)

Another experiment with experienced typists yielded equivocal results. Two offices of the same company were used, one a quiet office and one a noisy office. Output was slightly greater in the noisy office, but this tendency is vitiated by the fact that the typists in the quieter room were interrupted by about 25% more visitors. (33)

In another instance it is reported that moving the assembly department of a temperature regulator company from adjoining a noisy boiler shop to a quieter room resulted in lowering rejections at inspection from 75% to the low figure of 7%. The same reduction in noise through re-location of the department resulted in output jumping from 80 to 110 assembled units per unit of time. In another department removing the noise of a large ventilating fan resulted in output rising 12%. (34)

A 42% reduction in errors in the telephone operating room of a telegraph company followed lowering the noise level from 50 decibels to 35 decibels. An acoustical installation was used for the reduction in noise. There was also a 3% reduction in the cost per message. (36)

There are many other records from practical business experiments, but they cannot be included in this report since either permission is not forthcoming, or because the name of the firm or an individual as the responsible reporter cannot be used. All that can be reported is that in each of these reducable noise has been shown to be cutting into industrial output.

Summary

Acoustic stimuli have recordable effects on living organisms, varying from influencing the breathing rate to lowering of intelligence scores. That these effects are essentially biological and intrinsic in the organism itself is shown by their presence in functions which are removed from voluntary control as well as their presence in decerebrate animals. (35) In the main the effects are not beneficial, except in the possible case of musical stimuli where the findings cited are equivocal and inconsistent. It is hoped that this survey will give an impetus to researches along more practical lines, with stimuli under modern acoustical control and specifications.

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THE APPROXIMATE NETWORKS OF ACOUSTIC FILTERS

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Synopsis: The approximate equivalent electrical networks of acoustic filters are developed in this paper, from the lumped-constant approximation networks for electric lines. In terms of this network, design formulae have been developed for all single band pass filters. It is possible, from these formulae, to determine the physical dimensions of an acoustic filter necessary to have a given attenuation and impedance characteristic.

The original theory of acoustic filters given by Stewart[1] is based upon the representation of such filters by means of lumped constants in the form of a T network. More recently, the writer[2] has presented a theory of acoustic filters, showing that they are equivalent to a combination of electric lines. Lines, as an approximation, can be represented by networks with lumped constants, and hence an acoustic filter has a lumped-constant approximation network, which should represent the filter well at low frequencies. It is here shown that the network proposed by Stewart is a first approximation to the network of electric lines given in the former paper.[2,3] This first approximation represents the low pass filter well at low frequencies, but does not very adequately represent the band-pass filters. Accordingly, a second approximation is developed. All of the single band-pass filters have been analyzed and design formulae are given for them in terms of the second approximation network.

The Approximate Lumped-Constant Networks of Acoustic Filters

An acoustic filter, as developed so far, consists of a main conducting tube, and a side branch. In a symmetrical filter, the side branch is connected to the main conducting tube half-way between the two ends, as shown on Fig. 1. The type of filter obtained depends primarily on what type of side branch is used, the resonances of the latter determining the frequencies of maximum suppression.

The equivalent electrical circuit for an acoustic filter, was shown in a previous paper[2] to be two lines shunted by the impedance of the side branch. This representation is shown on Fig. 2. To obtain a lumped-

[1] Stewart, Phys. Rev., 20, pp. 528–551, 1922. Phys. Rev., 25, pp. 90–98, 1925.

[2] Mason, Bell System Technical Journal, 6, pp. 258–294, 1927.

[3] This fact has also been pointed out by Stewart "Journal of the Optical Society," July 1929, and by Lindsay, Phys. Rev., 24, pp. 652–655, 1929.

constant representation for this network, it is necessary first to consider the lumped-constant representation of a line, which is discussed below.

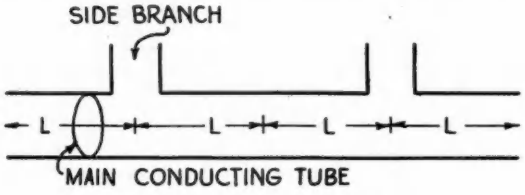


FIG. *Typical Acoustic Filter.*

A. Lumped-Constant Representation of a Line

In a previous paper,[2] it was shown that the propagation constant of a tube is given by the equation

$$P^2=\frac{-\omega^2}{c^2}\left[\left(1+\frac{P_0}{S}\sqrt{\frac{\gamma'^2}{2\omega\rho}}\right)-\frac{iP_0}{S}\sqrt{\frac{\gamma'^2}{2\omega\rho}}\right], \tag{1}$$

while the characteristic impedance is given by the expression

$$Z=\frac{\rho c^2 P}{j\omega S} \tag{2}$$

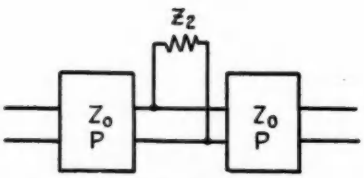


FIG. 2. *Electrical Equivalent of Acoustic Filter.*

In these equations ω is 2π times the frequency, C the velocity of sound, P_o the perimeter of the tube, S its area, ρ the density of the medium and γ'^2, a constant related to the viscosity, which for air has the value 4.25×10^{-4} in c.g.s. units.

A tube is the analogue of an electric line with distributed resistance, inductance, and capacity. No quantity corresponding to leakance is present. To determine the values of these quantities, use is made of the well known equations for a line

$$Z=\sqrt{\frac{R+j\omega L}{G+j\omega C}}\,; \quad P=\sqrt{(R+j\omega L)(G+j\omega C)} \tag{3}$$

[4] A. E. Kennelly "Artificial Electric Lines, 1917."
K. S. Johnson "Transmission Circuits for Telephone Communication, 1925"—Page 151.

where R, R, G and C are respectively the distributed resistance, inductance, leakance, and capacity of the line per unit length. Comparing (3) with (1) and (2), it is found that

$$R=\frac{P_0}{S^2}\sqrt{\frac{\gamma'^2\rho\omega}{2}}$$

$$L=\frac{\rho}{S} \tag{4}$$

$$C=\frac{S}{\rho c^2}$$

$$G=0$$

neglecting small correction terms. These are the equivalent distributed constants per unit length of the pipe expressed in acoustic impedance units.

The representation of lines with distributed constants by means of networks containing lumped constants has received considerable attention.[4] With three impedances, either the T or π network representation shown on Fig. 3, can be used.

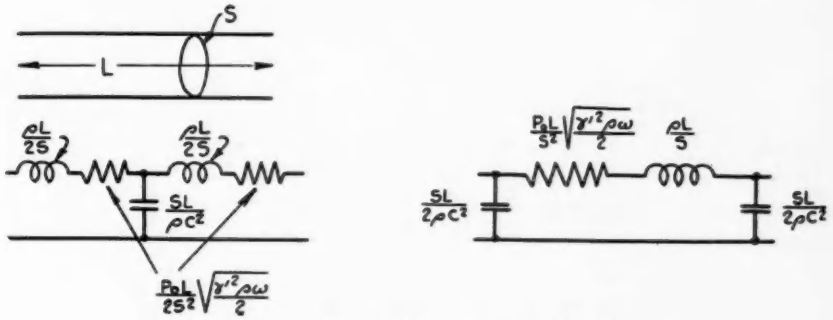


FIG. 3. *T or π Network Representations of a Line.*

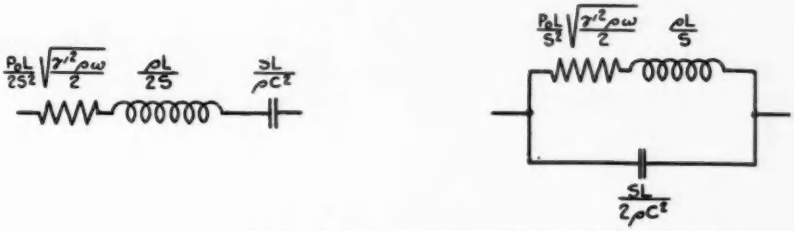


FIG. 4. *Representations for Open and Short Circuited Lines.*

The impedances of short or open circuited lines can be represented approximately by fewer elements than three. The first approximation for a short circuited line is an inductance and resistance equal to the sum of the distributed inductances and resistances of a line, while the first approximation for an open circuited line will be a capacity equal to the distributed capacities of the line. These approximations hold for very low frequencies. The second approximation for open and short circuited lines can be obtained with three impedances, as shown on Fig. 4. These representations follow directly from the T or π network representation shown on Fig. 3, by open or short circuiting the T and π networks, respectively.

B. Lumped-Constant Representation of an Acoustic Filter

In his theory of acoustic filters, Stewart has represented an acoustic filter by the network shown on Fig. 5, where Z_2 is the impedance of the side branch. Stewart has represented the side branch impedance, by

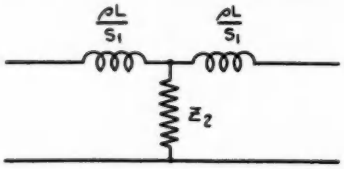


FIG. 5. *T Network Representation of Acoustic Filter.*

either one or two elements, depending on the side branch, and the main branch by a single inductance, equal to the sum of the distributed inductances of the tube. This corresponds to the first approximation of the representation of a line by lumped constants. This representation gives good results for the low pass filter, but does not represent, very adequately, the band-pass filters.

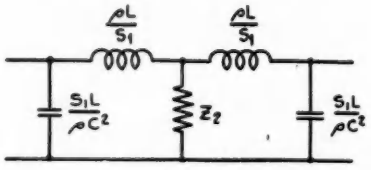


FIG. 6. *Second Approximation Network for Acoustic Filter.*

The best second approximation for an acoustic filter, employing two elements to represent the main conducting tube, is shown on Fig. 6. The main conducting tube is represented by an L network containing the total distributed capacity of the t[illegible] the shunt arm, and the

total distributed inductance of the tube in the series arm. The side branch impedance shunts the two L networks at their center.

The propagation constant and characteristic impedance of this structure are given by the expressions

$$\cosh P = 1 - \frac{2\omega^2 L^2}{c^2} + \frac{j\omega\rho L}{Z_2 S_1}\left(1 - \frac{\omega^2 L^2}{c^2}\right)$$

$$Z = \frac{\rho c}{S_1}\sqrt{\frac{1 + \dfrac{j\omega\rho L}{2Z_2 S_1}}{\left[1 - \dfrac{j\rho c^2\left(1 - \dfrac{\omega^2 L^2}{c^2}\right)}{\omega L S_1 (2Z_2)}\right]\left[1 - \dfrac{\omega^2 L^2}{c^2}\right]}} \qquad (5)$$

where S_1 is the area of the main branch.

If these equations are compared with those given in the former paper,[2] it is seen that they are approximately those obtained by taking the first two terms of the expansions of the trigonometrical functions. The

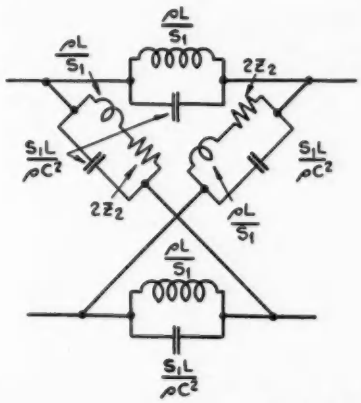


FIG. 7. *Lattice Network Representation for Acoustic Filter.*

characteristics of the filter are not very readily seen from equation (5), but can be readily found by transforming the network shown on Fig. 6, into the much more general lattice network shown on Fig. 7. That the network shown on Fig. 7 is the equivalent in characteristic impedance and propagation constant of that shown on Fig. 6, can readily be verified by substituting the impedances of the lattice network into the formulae for a lattice network

$$Z=\sqrt{Z_A Z_B}\ ;\ \cosh P=\frac{Z_B+Z_A}{Z_B-Z_A} \tag{6}$$

where Z_A is the impedance of one of the series arms, and Z_B that of one of the lattice arms. A lattice network has a pass band when the reactance of the series arm is of opposite sign to that of the lattice arm. When the reactances of the two arms have the same sign, an attenuation band results, while when the reactances of the two arms are equal, an infinite attenuation constant results, since here the lattice will be a balanced Wheatstone bridge.

For example, suppose that a side branch impedance, equivalent to an inductance and capacity in series, is used. The impedance of the lattice arm has two zero impedance points—one of which is at an infinite frequency—and two infinite impedance points—one of which is at zero frequency—as shown on Fig. 8. The impedance of the series arm

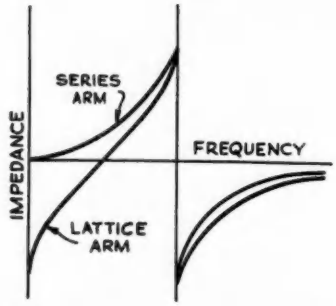


FIG. 8. *Frequency Impedance Characteristics.*

is that of an anti-resonant circuit, as shown on Fig. 8. There are two possible impedance characteristics for the series arm, in relation to the shunt arm, which will give a single band filter. One of these is obtained by letting the series arm have an infinite impedance when the lattice arm has a zero impedance, which results in a low pass filter. The second relation—which is that shown on Fig. 8—is obtained by letting the series arm have an infinite impedance when the lattice arm has an infinite impedance. The pass band is between zero frequency, and the frequency at which the lattice arm resonates.

In a similar manner, the other types of acoustic filters can be analyzed.

C. Side Branch Impedances

The possible types of side branches can be divided into two classes, those which are entirely enclosed, and those which are open to the air.

The first kind are characterized by a series capacity, while the second kind always have a shunt inductance.

One of the simplest side branch impedances is a short tube open on the end. The first approximation to this side branch is an inductance, as shown on Table I, #1, equal to the total distributed inductance of the tube. This approximation holds well if the product of the tube length by the frequency, is not too large. A longer tube, open on the end, can be represented by an inductance and capacity in parallel as discussed in Section A and shown on Table I, #2. A tube closed on the end can be represented by an inductance and capacity in series as shown on Table I, #4.

When these tubes are used as side branches, an additional factor comes in—an end correction. That is, the side branch must be considered as extending into the main branch for a distance proportional to the radius, because a motion of air in the direction of the side branch, occurs in the main branch. The value of this effect has been investigated by Rayleigh, who found that this effect can be calculated by increasing the length of the tube by a length equal to .785 times the radius. Another correction applies to an open ended tube, which has been determined experimentally as .57 times the radius. Hence the length of an open ended tube must be considered as

$$l' = l + (.785 + .57)r.$$

A straight tube can give all the combinations of side branch impedances, but one of its dimensions is necessarily limited, namely the area. For the area cannot become larger than the area of the main tube, since otherwise it could not be connected to the main tube. By using other types of side branches, this difficulty can at least be partially eliminated. For example, a concentric tube closed on the end is, to a first approximation, equivalent to an inductance and capacity in series, and it can be made to have a larger area relative to the main branch tube, than can the straight tube.

The choice of the forms of the structures to give the simplest impedance elements, is large. For example, Stewart represents a shunt inductance and capacity in parallel, by a concentric tube closed on the end, and a straight tube open on the end, joined together to the main conducting tube at a common point.[5] Other methods for representing two elements are shown on Table I. In these structures, the equivalent

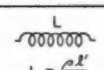
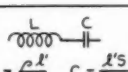
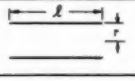
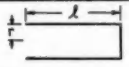
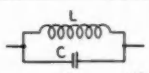
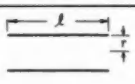
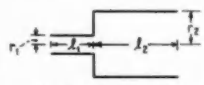
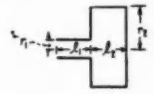
[5] See for example "Journal of the Optical Society" July 1929, Page 18.

length and equivalent areas have been calculated corresponding to these values for a straight tube. These elements have been calculated by calculating the impedances looking into the structures and taking the second approximations.

TABLE I

Equivalent electrical elements for acoustic side branch impedances.

TABLE-I

ELEMENT	 $L=\frac{\rho l'}{S}$	ELEMENT	 $L=\frac{\rho l'}{2S}$ $C=\frac{l'S}{\rho c^2}$
STRUCTURE No. 1		STRUCTURE No. 4	
VALUES OF CONSTANTS	$l' = l + 1.355r$ $S = \pi r^2$	VALUES OF CONSTANTS	$l' = l + .785r$ $S = \pi r^2$
ELEMENT	 $L=\frac{\rho l'}{S}$ $C=\frac{l'S}{2\rho c^2}$	STRUCTURE No. 5	
STRUCTURE No. 2		VALUES OF CONSTANTS	$l' = \sqrt{\left[\frac{\text{Log}\left(\frac{r_1+r_2}{2r_1}\right)+\frac{.46}{t}}{T}\right]\left[(r_2^2-r_1^2)T+\frac{r_1^2 t}{2}\right]}$ $S = \pi\sqrt{\frac{\left[(r_2^2-r_1^2)T+\frac{r_1^2 t}{2}\right]}{\frac{\text{Log}\left(\frac{r_1+r_2}{2r_1}\right)+\frac{.46}{t}}{T}}}$
VALUES OF CONSTANTS	$l' = l + 1.355r$ $S = \pi r^2$		
STRUCTURE No. 3	 $l_1' = l_1 + .785\,r_1 \quad S_1 = \pi r_1^2$ $l_2' = l_2 + .57\,r_2 \quad S_2 = \pi r_2^2$	STRUCTURE No. 6	 $l_1' = l_1 + .785\,r_1 \quad S_1 = \pi r_1^2$ $l_2' = l_2 \quad S_2 = \pi r_2^2$
VALUES OF CONSTANTS	$l' = \sqrt{\frac{2\left[S_1(l_2'^3+3l_1'^2l_2')+\frac{S_2^2}{S_2}(3l_1'l_2'^2)+l_1'^3S_2\right]}{3(S_2l_1'+l_2'S_1)}}$ $S = \sqrt{\frac{2S_1^2S_2[S_1S_2(l_2'^3+3l_1'^2l_2')+3S_1^2l_1'l_2'^2+S_2^2l_1'^3]}{3(S_2l_1'+l_2'S_1)^3}}$	VALUES OF CONSTANTS	$l' = \sqrt{\frac{2[l_1'^3+3l_1'l_2'^2)S_1S_2+3S_1^2l_1'^2l_2'+l_2'^3S_2^2]}{3S_2[S_1l_1'+S_2l_2']}}$ $S = \sqrt{\frac{3S_2(S_1l_1'+S_2l_2')^3}{2[l_1'^3+3l_1'l_2'^2)S_1S_2+3S_1^2l_1'^2l_2'+l_1'^3S_2^2]}}$

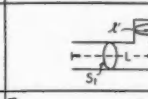
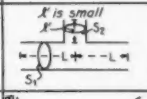
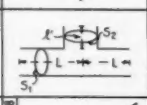
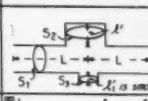
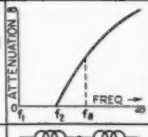
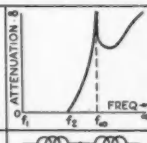
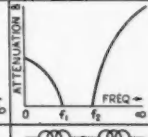
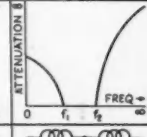
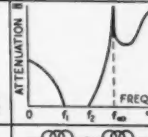
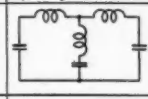
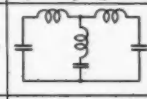
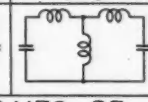
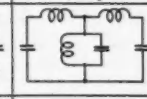
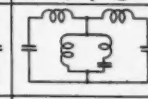
D. Design Formulae for Acoustic Filters

Using the side branch impedances shown in Table I, in the lattice network shown by Fig. 7, the resulting characteristics can readily be

obtained. A large number of multiband characteristics can be secured by using various combinations of side branches, but only five single band filters (to the degree of approximation considered here) have been found. The attenuation characteristics of these filters and the design

TABLE II

Design formulae for acoustic filters.

TABLE-II					
NO.	1	2	3	4	5
STRUCTURE					
ATTENUATION CHARACTERISTIC					
EQUIVALENT ELECTRICAL STRUCTURE					
		VALUES OF			
f_1	0	0	$\frac{C}{2\pi L\sqrt{1+\frac{2L'S_1}{LS_2}}}$	$\frac{C}{2\pi L}\sqrt{1-\sqrt{1-\frac{2L^2}{L'^2}}}$	$\frac{C}{2\pi L\sqrt{1+\frac{L'S_2}{LS_3}+\frac{L'S_1S_2}{S_3L^2}}}$
f_2	$\frac{C}{2\pi\sqrt{\left(\frac{L}{S_1}+\frac{L'}{S_2}\right)\frac{L'S_2}{2}}}$	$\frac{C}{2\pi}\sqrt{\frac{1+\frac{L'S_2}{2LS_1}}{\frac{LL'S_2}{S_1}\left(1+\frac{L'S_1}{LS_2}\right)}}$	$\frac{C}{2\pi L}$	$\frac{C}{2\pi L}\sqrt{1+\sqrt{1-\frac{2L^2}{L'^2}}}$	$\frac{C}{2\pi\sqrt{\frac{L'^2}{2}+\frac{LL'S_1}{LS_3/S_2+L'S_1/S_2}}}$
f_∞	∞	$\frac{C}{2\pi L}\sqrt{\frac{1+\frac{L'S_2}{2LS_1}}{1+\frac{2L'S_1}{LS_2}-\frac{LS_1}{L'S_2}}}$	∞	∞	$\frac{C}{2\pi L}$ and ∞
f_δ	$C/2\pi L$				
Z_0	$\frac{\rho C}{S_1\sqrt{1+\frac{L'S_2}{2S_1L}}}$	$\frac{\rho C}{S_1}\sqrt{\frac{1}{1+\frac{L'S_2}{2S_1L}}}$	$\frac{\rho C}{S_1}\sqrt{\frac{1+\frac{2L'S_1}{LS_2}}{\left[1+\frac{2L'S_1}{LS_2}-1\right]^2}}$	$\frac{\rho C}{S_1}\sqrt{\frac{L'^2}{2(L'^2-2L^2)}}$	$\frac{\rho C}{S_1}\sqrt{\frac{1+\frac{2LS_1}{LS_3}\left(\frac{L'^2}{2LL'}+\frac{L'S_1/L}{LS_3/S_2+L'S_1/S_2}\right)}{\frac{L'^2}{2S_1^2}+\frac{L'S_1/L}{LS_3/S_2+L'S_1/S_2}\left(1+\frac{LS_3}{S_3L'}\right)}}$
Z	$\frac{\frac{\rho C}{S_1}\sqrt{\frac{f_2^2}{f_\infty^2}\left(1-\frac{f^2}{f_2^2}\right)}}{1-\frac{f^2}{f_\infty^2}}$	$\frac{\frac{\rho C}{S_1}\sqrt{\frac{f_2^2}{f_\infty^2}\left(2-\frac{f_2^2}{f_\infty^2}-2\sqrt{1-\frac{f_2^2}{f_\infty^2}}\right)}}{1-\frac{f^2}{f_2^2}}$	$\frac{j\frac{\rho C}{S_1}\sqrt{\frac{f^2}{f_2^2}}}{\sqrt{\left(1-\frac{f^2}{f_2^2}\right)\left(1-\frac{f^2}{f_\infty^2}\right)}}$	$\frac{j\frac{\rho C}{S_1}\sqrt{\frac{f^2}{2f_2^2}\left(1+\frac{f_2^2}{f_1^2}\right)}}{\sqrt{\left(1-\frac{f^2}{f_1^2}\right)\left(1-\frac{f^2}{f_2^2}\right)}}$	$\frac{j\frac{\rho C}{S_1}\sqrt{\frac{f^2f_2^2}{f_1^2f_\infty^2}\left(1-\frac{f_1^2}{f_2^2}\right)}}{\sqrt{\left(1-\frac{f^2}{f_1^2}\right)\left(1-\frac{f^2}{f_\infty^2}\right)^2}}$
		DESIGN FORMULAE			
L	$\frac{C}{2\pi f_\delta}$	$\frac{C}{2\pi f_\infty\left[1-\sqrt{1-(f_2^2/f_\infty^2)}\right]}$	$\frac{C}{2\pi f_2}$	$\frac{C}{\pi\sqrt{2(f_2^2+f_1^2)}}$	$\frac{C}{2\pi f_\infty}$
S_1	$\frac{f_1\rho C}{Z_0f_\delta}$	$\frac{\rho Cf_\infty\left[1-\sqrt{1-(f_2^2/f_\infty^2)}\right]}{Z_0f_2}$	$\frac{\rho Cf_\delta}{Z_0(f_2-f_1)}$	$\frac{\rho C(f_2^2+f_1^2)}{2Z_0(f_2^2-f_1^2)}$	$\frac{\rho Cf_2f_\infty}{Z_0(f_\infty^2-f_1f_2)}$
L'	$\frac{\sqrt{2}C}{2\pi f_\delta}$	$\frac{\sqrt{2\left(1-\frac{f_2^2}{4f_\infty^2}-\sqrt{1-\frac{f_2^2}{f_\infty^2}}\right)}}{2\pi f_\infty\left[1-\frac{f_2^2}{2f_\infty^2}-\sqrt{1-\frac{f_2^2}{f_\infty^2}}\right]}$	—	$\frac{C\sqrt{f_2^2+f_1^2}}{2\pi f_1f_2}$	$\frac{C}{\sqrt{2}\pi f_\infty}$
S_2	$\frac{(f_2^2-f_1^2)\rho C\sqrt{2}}{f_1f_\delta Z_0}$	$\frac{2\left[f_2^2-f_\infty^2\left(1-\sqrt{1-\frac{f_2^2}{f_\infty^2}}\right)\right]}{Z_0f_2\sqrt{\rho C^2\left(1-\frac{f_2^2}{4f_\infty^2}-\sqrt{1-\frac{f_2^2}{f_\infty^2}}\right)}}$	—	$\frac{4\rho Cf_1f_2(f_2^2+f_1^2)^{3/2}}{Z_0(f_2^2-f_1^2)^2(f_2-f_1)}$	$\frac{\rho C\sqrt{2}f_2f_\infty(f_\infty^2-f_2^2)}{Z_0(f_\infty^2-f_1f_2)(f_2^2-f_1^2)}$
$\frac{L'}{S_2}$	—	—	$\frac{Z_0(f_2^2-f_1^2)(f_1+f_2)}{4\pi\rho f_1^2f_2^2}$	—	—
$\frac{L'_1}{S_3}$	—	—	—	—	$\frac{Z_0(f_2^2-f_1^2)(f_\infty^2-f_1f_2)}{4\pi\rho f_2f_1^2f_\infty^2}$

formulae for them are shown on Table II, In designing a filter, it is usual to obtain the dimensions in terms of the singular frequencies which determine the action of the filter. One other parameter appears, Z_0, which represents the characteristic impedance of the filter at the mean frequency of the band i.e. $f_m = \sqrt{f_1 f_2}$. It is usual to match, approximately, the impedance terminations of the filter to the value Z_0.

All of these filters have been calculated for side branch tubes, of constant cross section but any of the other side branches shown on Table I can be used by employing the equivalent values of l' and S shown there.

The frequency f_a appearing in the filter #1 has no significance for the attenuation constant. It determines the frequency at which the characteristic impedance equals infinity. Considering the loss caused by inserting the filter between two impedances equal approximately to Z_0, an additional loss occurs at the frequency f_a, due to a mismatch of the impedance of the filter and the terminating impedances. Filter #4 of Table II is similar to #3 except that it has twice the attenuation constant. It is then equivalent to two sections of the #3 filter.

DESCRIPTION AND DEMONSTRATION OF AN ARTIFICIAL LARYNX

By R. R. Riesz
Bell Telephone Laboratories

In the production of the various voiced speech sounds, the initial source of sound is the human larynx, more particularly the vocal cords. The sound generated by the larynx is rich in harmonic content and before it is radiated into the external atmosphere its quality is modified by the resonating action of the cavities of the pharynx, mouth and nose. The different combinations of the shapes of these cavities give rise to the various voiced sounds of speech. The left-hand diagram of Figure 1 shows the location of the larynx in a normal individual. It is situated at the top end of the trachea or wind pipe just behind the base of the tongue, the thyroid cartilage of the larynx causing the prominence

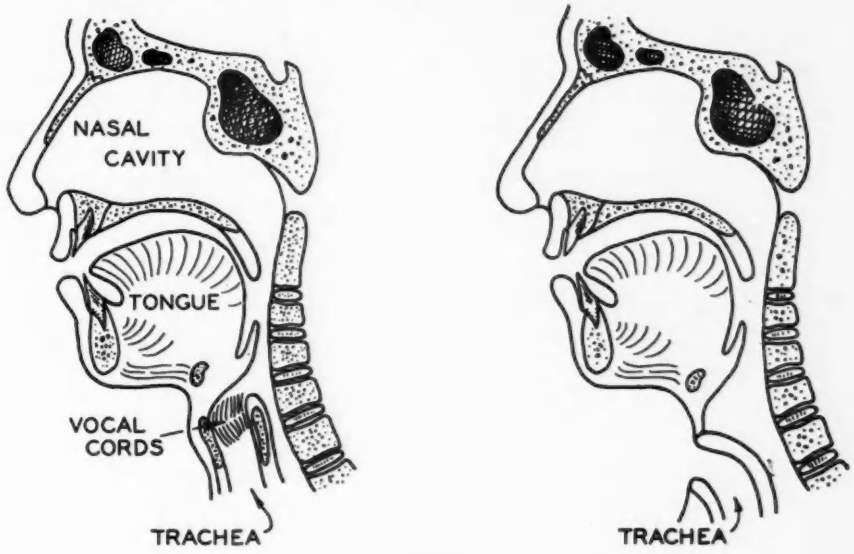


Fig. 1.

at the front of the throat known as the Adam's Apple. During breathing the vocal cords, a pair of fibrous lips, are drawn to the sides of the larynx leaving a free air path to and from the lungs by way of the trachea. In the act of phonation the vocal cords are drawn close together leaving a slit a few tenths of a millimeter wide between them. This slit is known as the glottis. Then the pressure of the air in the lungs is in-

creased and a stream of air is made to flow through the glottis. This stream of air sets the vocal cords into vibration which in turn imposes periodic fluctuations on the flow of air through the glottis. These fluctuations are the sound waves that are modified by the vocal cavities, tongue, and teeth into the sounds of articulate speech.

In the United States there are several hundred people who because of malignant disease have had their larynx removed by a surgical operation. The right-hand diagram of Figure 1 shows a crude representation of the state of things after the removal of the larynx. The lungs

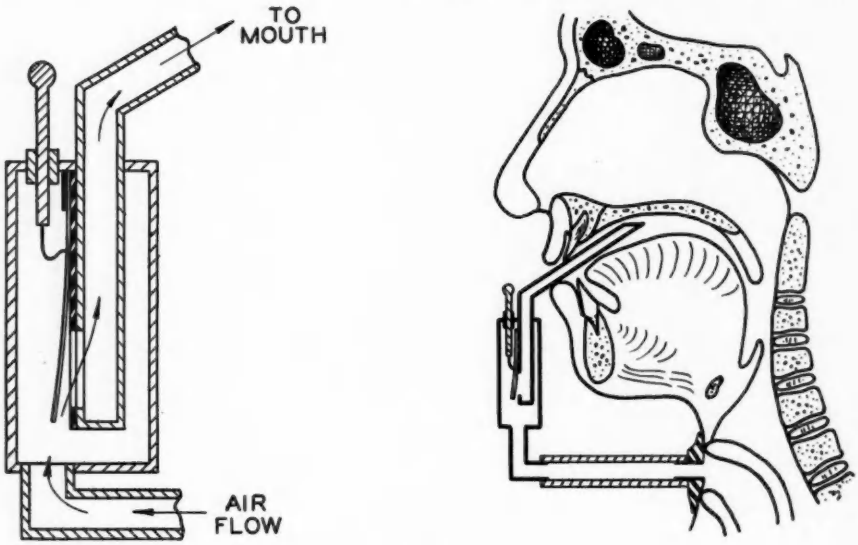


FIG. 2.

are entirely disconnected from the mouth and throat, the trachea being terminated directly in a small opening in the front of the throat. It is through this opening that breathing takes place and, of course, speech of any kind is impossible except with some device such as an artificial larynx. A recently developed model of an artificial larynx is shown on Figure 2. It consists of a cylindrical case divided into two compartments by a cylindrical partition having one flat side which is covered with a sheet of rubber. Air can pass from one compartment to the other through an elongated hole cut through the flat side of this partition. A thin reed of spring metal, clamped at one end, is held in position over this hole. As air is blown through the hole it sets this reed into vibration which periodically varies the flow of air through the larynx and so superposes a train of sound waves on the steady flow of

air. For a satisfactory artificial larynx the quality of the sound generated must be a fair duplication of that produced by the normal vocal cords. It might be pointed out that the series of harmonics of the sound produced is not that corresponding to the series of overtones produced by a clamped-free reed, since the reed as used merely plays the part of a regulator in periodically augmenting and diminishing the flow of air. Harmonic analyses of the sound produced showed a complete series of harmonics present. The fundamental frequency of the artificial voice can be changed by means of the sliding clamp which when it is moved downward reduces the effective length of the vibrating reed and so raises the frequency of oscillation. With the clamp set at a definite position the fundamental pitch can be further changed through several semi-tones by using a stronger or weaker current of air. This enables the user readily to inflect his voice.

In use, the input tube of the artificial larynx is connected to the breathing opening by means of a rubber tube and a coupling pad held against the front of the neck. The sound generated by the steady flow of air from the lungs is introduced into the mouth by a flexible rubber tube. In speaking the user goes through a series of motions and muscular adjustments that are practically the same as those used before the removal of the larynx. The action of the vocal cavities in converting the sound produced by the artificial larynx into articulate speech is identical with that when a normal larynx is used, with the exception that in the former case the sound is introduced into the mouth while normally it is introduced at the base of the pharynx. This difference is apparently of secondary importance, for people using the artificial larynx constantly often attain a high degree of proficiency in producing intelligible speech.

A normal person can learn to use the artificial larynx by supplying air to it from a hand bellows instead of from the lungs. I have here such a bellows. This is the sound produced by the artificial larynx and corresponds to the tone rich in harmonics produced by the vocal cords (demonstration). By shaping the vocal cavities to pronounce the vowel *a* as in father and introducing this sound into the mouth a recognizable *a* is produced (demonstration). Similarly, for the vowel *ō* (demonstration). With a small amount of practice all the speech sounds used in English can be produced with a fair degree of proficiency, the most difficult being the semi-vowel *ng*. This is probably due to the importance of the nasal cavity in producing this sound and the fact that the tongue is held in such a position as practically to close the opening be-

tween the mouth and the pharynx. The sound being introduced into the mouth, very little acoustic effect is produced in the nasal cavities.

Using the artificial larynx, it is also possible to carry on a conversation: "Good afternoon members of the Acoustical Society of America. You see not only can isolated speech sounds be produced with the artificial larynx but a fairly intelligible conversation can be carried on. When first learning to talk with this device it is difficult to secure continuity to the flow of speech, some of the consonants being particularly troublesome. However, the use of a few simple tricks soon overcomes

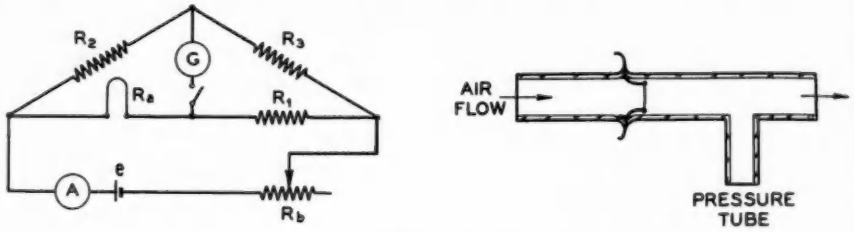


FIG. 3.

this. The effect of raising the pitch of the larynx can be shown. This particular larynx has a range of about an octave and I am now speaking at a pitch suitable for a woman's voice." (Spoken with artificial larynx.)

The artificial larynx when being used by a person whose normal larynx has been removed furnishes a rare opportunity for simultaneously observing the excess pressure in the lungs and the rate of explulsion of air when speaking—a thing that cannot be done for a normal individual. Figure 3 shows the apparatus used. A short piece of $\frac{1}{2}''$ glass tubing was placed in the air line connecting the trachea opening and the artificial larynx. A small side tube leading from this to a water manometer enabled the excess pressure in the lungs to be determined. A fine .0015″ platinum wire was sealed into the tube so that it was stretched normal to the axis of the tube. It was used in the circuit shown as a hot wire anemometer to determine the rate of expulsion of air from the lungs. The anemometer, represented by R_A, was used as one arm of a Wheatstone's bridge. With the current flowing through the fine platinum wire, the anemometer was shielded from air currents so that no air flowed through it and the bridge was balanced. Any air flow through the anemometer reduced the temperature of the platinum wire and so unbalanced the bridge. A calibration of rate of air flow in terms of the current through the microammeter reading unbalance of

the bridge was made. With a steady flow of air through the tube the indicating meter reached its equilibrium value in about 3.5 seconds.

Using this instrument 10 artificial larynges of different degrees of difficulty of operation were given to a person whose normal larynx

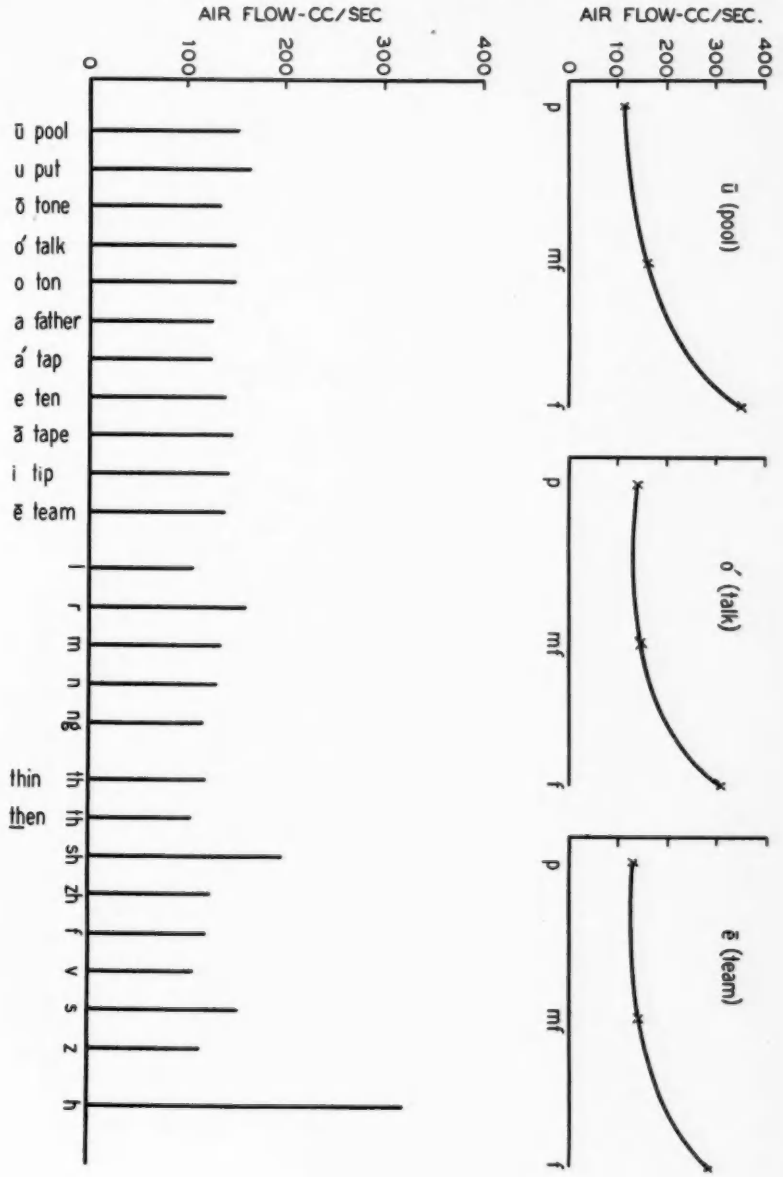


FIG. 4.

had been removed. For each instrument, readings of excess pressure in the lungs and rate of expulsion of air were made and a note made of the effort required to operate the larynx. Finally, one artificial larynx was singled out that in the opinion of the user seemed to take the same effort to operate as normal speech had before the removal of the normal larynx. If this is considered representative, the excess pressure in the lungs when producing a sustained vowel at a conversational intensity is that corresponding to about 4 mm of mercury or .005 atmospheres and the rate of expulsion of air is about 125 cc per second. This means that in producing a sustained vowel the lungs do work at the rate of approximately .06 watts in blowing air through the glottis. This figure may be compared with the mean rate at which the heart does work in pumping blood through the circulatory system; this is about 4 watts.[1] The power used by the lungs in speaking is only about 1.5% of this. The rate of radiation of acoustic energy from the lips of a speaker when producing a sustained vowel at a conversational intensity is estimated as about 100 microwatts. Therefore, of the energy used by the lungs in operating the vocal cords only about 0.17% is converted into useful acoustic energy.

A second hot wire anemometer was used with the above circuit to measure the rate of expulsion of air by normal persons in producing sustained sounds. The air expelled from the mouth and nose was collected by an air tight mask and passed through the anemometer. The mask produced no intelligible impairment to speaking. Measurements were made on the rate of expulsion of breath in sustaining the English vowel sounds at three intensities; very weak, conversational and very loud. Figure 4 shows three representative curves, the average of three male observers intoned had about 120 cps. Measurements with a volume indicator showed that these intensities were approximately in the ratio, 1:100:10,000.

The values of rate of expulsion of air ranged from about 110 to 300 cc per second. Weak and conversational intensities take about the same amount of air while the loud intensity takes considerably more. The lower chart shows the volume of air expelled per second from the lungs in producing various sustainable English speech sounds at a conversational intensity. It is noticed that *h* uses more than twice the amount of air than other sounds do but this is to be expected since the vocal cords are wide open when producing this sound. For the fricative

[1] Howell, Text Book of Physiology.

consonants each unvoiced consonant uses more air than its voiced mate; probably the added resistance of the glottis reduces the air flow in the latter case. The differences between the vowels are not great enough to be significant.

Without measurements of the corresponding pressures in the lungs, these data are incomplete. But if, as a not-unlikely speculation, it be assumed that in producing the sustained vowels the pressure in the lungs is proportional to the rate of expulsion of air, the powers used in sustaining the vowels at the three intensities measured are about as 1:1:8. Since the intensities of the corresponding acoustic outputs are in the ratio 1:100:10,000 the efficiency of the larynx as a generator of sound waves is greatest at high intensities, being 1200 times what it is at low intensities and about 12 times what it is at conversational intensities. The explanation of these facts would probably necessitate a detailed study of the dimensions of the glottis and the muscular adjustments of the parts of the larynx when producing sounds at various intensities.

MAY 22, 1929.

ACOUSTICS OF THE SALT LAKE TABERNACLE

By Wayne B. Hales
Weber College

Introduction

During the summer months from 500 to 2000 tourists from all over the world visit the Salt Lake tabernacle daily. Guides are furnished who conduct them through the building, explaining as they go the method of and difficulties attending its construction, its dimensions, seating capacity, etc., and conclude by giving them a demonstration of some of its acoustic properties. An attendant drops a pin on a wooden rail, drops it in his stiff hat, rubs his hands, and whispers toward and away from the visitors who are in the balcony at the rear of the building nearly 200 feet away. All of these demonstrations are heard distinctly. The result is that the "Mormon" tabernacle has a very extended reputation of being one of the most acoustically perfect buildings in the world. Being a "whispering gallery" of eminence, however, is not sufficient evidence to determine the perfectness of the acoustics of any building, but rather the hearing must be good in all positions in the auditorium and under all conditions as to the nature of the source emitting the sound.

Because of the widespread reputation of the Salt Lake tabernacle as an auditorium of extraordinary acoustical qualities, and because of its historical significance, the author became interested in carrying out certain acoustical experiments in the building.

An attempt was made therefore to—

I. Determine the variation of reverberation with a variation of:
   (a) Absorption power,
   (b) Position of observer in auditorium,
   (c) Pitch,
   (d) Intensity;

II. Determine to what extent and under what conditions echoes are produced in the auditorium;

III. Determine the distribution of the intensity of sound throughout the auditorium;

IV. Determine the per cent of words articulated in the building under varying conditions.

Method

The method used on all reverberation measurements was essentially that of Sabine. The source of sound was the tabernacle organ itself which could be made to operate, by a proper adjustment of stops, 1, 2, 3, or 4 pipes (of the same pitch and practically of the same quality and intensity) by pressing but one key. The key used in the main was Middle C, vibrating 261 times per second. A determination of the period of reverberation consisted of an average of 20 observations. To eliminate the interfering effect of extraneous noises all determinations were made from 1 to 5 o'clock in the morning. The agreement of determinations taken by one observer on the same date, those taken on different dates, and those taken by a different observer are given in Table no. 1.

Table 1.

First:—Agreement of Observations Obtained at One Setting

Determination	Average Time for 20 Observations	Departure from Mean
1	4.62 sec.	−.05 sec.
2	4.62 "	−.05 "
3	4.68 "	+.01 "
4	4.68 "	+.01 "
5	4.80 "	+.13 "

Average deviation from mean .05 sec. Maximum deviation from mean .13 sec. Probable error less than .05 sec.

Second:—Agreement of Observations Obtained on Different Nights by the Same Observer and Under Similar Conditions

Date	Reverberation
June 23	4.67 sec.
June 26	4.73 "
July 11	4.66 "
July 14	4.75 "

Third:—Agreement of Independent Observations Made by Different Persons

Date	Name of person	Reverberation
July 29	J. J. Toronto	4.69 sec.
		4.68 "
		4.62 "

Reverberation and Absorption Power

By making use of Sabine's formula

$$at = k, \tag{1}$$

(a) being the total absorbing power of walls etc., t the period of re-

verberation, and k a constant depending upon the volume of the building according to

$$k = .164V, \qquad (2)$$

and by opening the many doors and windows in the building we had means of determining (a) and (k). If (x) = the number of square meters of open window space and therefore their respective absorbing power, then,

$$(a+x)t = k' \qquad (3)$$

x was made to vary from 0 to 137.7 sq. meters and t varied from 5.07 sec. to 4.30 sec. respectively. By substituting these various values of x and t in equation (3) and solving for (a) and k' we obtain an average for

$$a = 629 \qquad k' = 3190.$$

The detailed results are given in Table No. 2.

TABLE 2.

ABSORPTION AND REVERBERATION TABLE

Determination	No. of doors open	Area of openings	Absorbing power	Absorbing power corrected x	Reverberation in seconds t
1	0	0	0	0	5.07
2	2	13.4	13.4	12.6	5.05
3	4	26.8	26.8	25.2	4.92
4	6	39.7	39.7	37.3	4.78
5	8	51.5	51.5	48.4	4.70
6	10	62.4	62.4	58.6	4.61
7	12	72.3	72.3	67.9	4.42
8	14	82.0	82.0	77.0	4.53
9	14 doors				
	27 windows	137.7	137.7	132.2	4.30
10	27 windows	55.5	55.5	54.0	4.66

An opportunity was given during an organ recital to measure just how the reverberation would be affected by an audience in the tabernacle. With 1004 people in the balcony, all windows open, $(x = 55.5)$ and at noon competing with extraneous noises, $(t = 3.20$ sec.), equation (3) now becomes

$$(684+W)3.20 = 3190.$$

W = 313 open window units. This gives absorbing power per person = .31 and absorbing power per square meter of audience = .73 as compared with Sabine's values of .44 and .96 respectively.

Reverberation and Position of Observation

Determinations of reverberation were made in 11 different positions in the building: in the balcony, under the balcony and in the main body of the auditorium. All determinations agreed within .15 sec. to the average period of 4.76 sec., which shows that the period of reverberation is practically independent of the position in the tabernacle of the auditor.

Reverberation and Pitch

After an abrupt cessation of a chord on an organ, piano, or from an orchestra in an auditorium one notices the changing quality of the residual sound as it decays to minimum audibility. This is due to three causes: First, the materials in the building have a selective absorption

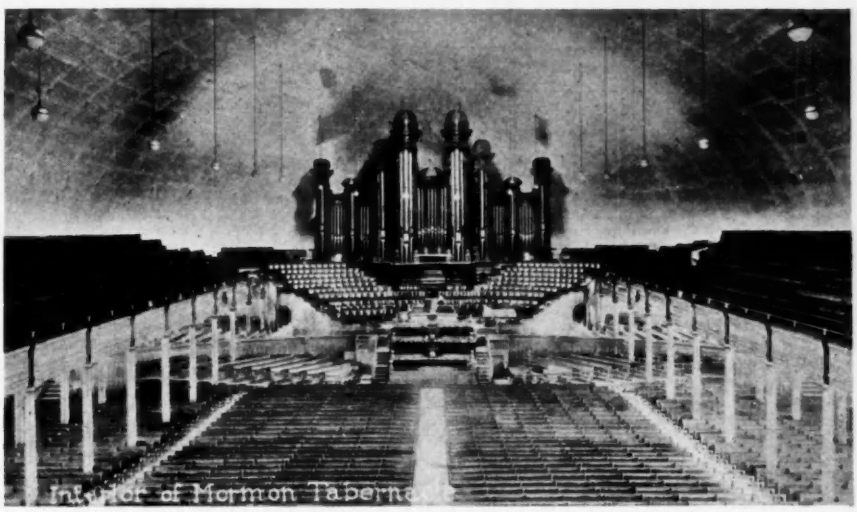


Fig. 1. *Interior of Mormon Tabernacle.*

characteristic. This tends to eliminate by absorption some notes much more rapidly than others, leaving the prolonged notes of a purer tone quality. Second, the intensity of the sound energy varies with pitch. Lord Rayleigh gives the formula for the energy per cu. cm. as $E = 2\pi^2 d n^2 a^2$ where d = density of medium, n = frequency and a = amplitude. Third: the sensitivity of the ear varies with pitch. To show the

individual effect of any one of these three factors could not be attempted in this experiment. The results, therefore, show only the combined effect.

Determinations were made for 16 different notes, all in the same set of pipes and blown with the same pressure, namely, .54 pounds per square inch. The period of reverberation varied from $t=1.63$ sec. for C_4, 16 v.p.s. to a maximum of $t=5.21$ sec. for F, 348 v.p.s., thence down to $t=0.97$ sec. for C^v, 8,368 v.p.s. These results show that the duration of residual sound under these conditions varies greatly with pitch, being relatively short for the high and low frequencies as compared with the time for the note F, 348 v.p.s. They are shown graphically in Figure 2.

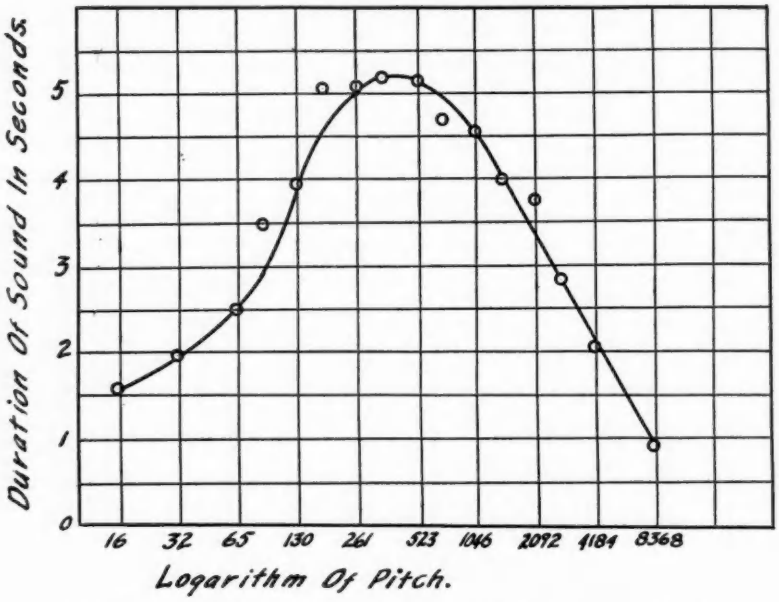


FIG. 2. *Curve showing variation of reverberation with variation of pitch.*

REVERBERATION AND INTENSITY

This part of the experiment was attended with some uncertainties. To obtain accurate results one should have four exactly similar pipes which when blown by the same pressure will give notes of equal intensity, and when blown together will give intensities 2, 3 and 4 times greater than one alone, providing the vibrations of one are not affected by that of another. Not having four similar pipes giving notes of equal intensities and quality on the Tabernacle organ, the writer used four notes which were as much alike as could be obtained. The pipes were

all of the same pitch, Middle *C* 261 v.p.s., and blown by an air pressure of .54 lbs. per sq. in.

By determining the duration of residual sound when first one pipe is blown alone and then two, three, and four pipes together, some very interesting facts may be determined. They are:

(1) The initial intensity (I_1) of the sound in terms of the intensity of minimum audibility (i').

(2) The constant of decay (A) under different absorbing conditions.

(3) The mean free path of sound between reflections.

(4) The determination of the volume of the building which will serve as a check on actual measurement of the volume.

In making these determinations use was made of the following formulae taken from Sabine's work:

$$I_1 = i'e^{At} \qquad (4) \qquad\qquad At_n = \log \frac{nI_1}{i'}, \qquad (5)$$

$$A = \frac{\log n}{t_n - t_1} \qquad (6) \qquad\qquad A = \frac{va}{pS} \qquad (7)$$

$$p = \frac{v.164V}{AST} \qquad (8) \qquad\qquad k = \frac{k'T}{t} \qquad (9)$$

where I_1 = intensity of sound at the stopping of the sounding pipe

i' = intensity of minimum audibility
I = 1,000,000 i'
A = constant of decay
n = number of pipes in operation
v = velocity of sound, 343 m. per sec. at 22°C.
V = volume of tabernacle = 35,300 c. m. from direct meas.
S = absorbing surface = 9,350 sq. meters from direct meas.
p = mean free path between reflections
t_n = period of reverberation for n pipes
T = period of reverberation for I = 1,000,000 i'
k = hyperbolic parameter for I = 1,000,000 i'

Observations and computations are given in Table no. 3.

Sabine proved experimentally that in irregularly shaped rooms, the mean free path is proportional to the cube root of the volume. Two rooms in the Boston Public Library of which he makes record have volumes 2,140 cu. meters and 63.8 cu. meters and mean free paths 7.8

TABLE 3.

	Square Meters of Open Windows			
	$W=0$	$W=54$	$W=132$	Average
*t_1=	4.86	4.59	4.44 sec	
t_2	5.67	5.17	5.02 "	
t_3	5.83	5.34	5.11 "	
t_4	5.97	5.43	5.30 "	
A (average) from (6)	1.23	1.43	1.53	
I_1	$437i'$	$702i'$	$898i'$	
T for $I=1{,}000{,}000i'$	9.37	8.05	7.53 sec	
k for $I=1{,}000{,}000i'$	6,040	5,600	5,400	5,680
p from (7) in meters	18.8	17.6	18.2	18.2
p from (8) in meters	18.4	18.5	18.4	18.4
V from $k=.164V$	36,800	34,200	32,900	34,630
V from direct measurements=				35,300

* t_1 was slightly different for each of the 4 pipes used.

and 2.27 meters, respectively. The mean free path in the tabernacle determined by comparison with these give 19.9 and 18.6 meters. This becomes an interesting check upon measurements recorded in this paper.

ECHOES IN THE TABERNACLE

If the source of sound in a room is a short, sharp impulse, it will spread out in all directions as a group of waves. If the room is a large one and if an observer is standing in the center, he will very likely hear two or three repetitions of the same impulse, the first one coming direct from the source, the others being a reflection and double reflection from the walls and ceiling. If he be seated in a position where the difference in path of the direct impulse and the reflected impulse is less than 75 feet he will perceive the two portions of the wave in the form of a single sensation of hearing, each louder than the effect which either part would have produced alone. In this case, instead of hearing an echo, the original impulse was reinforced. If the wall is curved the effect is generally either a more pronounced echo or increased reinforcement. In an auditorium the size and shape of the tabernacle one would expect both of these phenomena to be present. This is why a listener can hear so much better in the rear end of the balcony than in the middle of the main floor where he is from 50 to 100 feet closer to the speaker. In the balcony he hears the original impulse reinforced by reflections from

behind, above and on both sides. In the center of the building and for some distance behind a direct impulse of one syllable reaches the ear of the listener at the same time that the reflected impulse of the syllable that just preceded. The result is confusion or strained attention which becomes wearisome to the extreme if long maintained.

An effort was made to determine to what degree successive echoes were produced in the tabernacle. The source of sound was a highly damped clap produced by the impact of a block of wood and a block of iron. An attendant of the building operated the source at the main pulpit, while the author made observations at every point on the main floor and in the balcony. Nowhere were successive echoes perceived except down the center aisle, from a point very near the center of the floor to a point 30 feet back. In this region three audible successive echoes were heard. Fifteen feet on either side of this region there existed a phenomenon which could hardly be called echo but could be better described as a throbbing reverberation. It is in this region, probably 50 feet square, where hearing is attended with greatest difficulty because of the interference produced by successive reflections or echoes. Other places were equally bad but they are a result of low intensity rather than interference due to echo.

When the source of sound was carried by the observer, four points in the balcony symmetrical with respect to the center line of the building were strikingly singular. At these points as many as 14 distinct successive echoes could be heard all within 3 seconds after the clap of sound was made. This phenomenon is not a serious acoustical defect of the building since a source of sound is never at these points.

Distribution of Intensity of Sound in the Tabernacle

The method used in this part of the experiment was one suggested by Dr. Harvey Fletcher and used on similar experiments in the Bell Telephone laboratories. The source of sound was a vacuum tube oscillator, vibrating 806 times per second, located at the main pulpit. Two telephone receivers, mounted on a head support and held $1\frac{1}{2}$ inches from the ear by a wire cage, were carried to all parts of the building. These head phones were shunted across a potentiometer arrangement by which the current in them and likewise the intensity of the sound emitted by them could be varied to any degree desired. The current was first sent through the stationary receiver and then

through the head receivers. The potentiometer was adjusted until the intensities were the same from both sources. When this balance is made it can be shown that the intensity of the sound produced varies directly as the square of the shunt resistance. The unit of intensity was taken as the intensity of sound when the head phones were one meter from the pulpit receiver. Use was made throughout of the formula:

$$I_n = \left(\frac{R_n}{R_0}\right)^2$$

where R_o is the balance resistance one meter from the source R_n the balance resistance at ony other station in the auditorium and I_n is the corresponding intensity. Observations were made at 37 different positions in the building. These showed variations in intensity from 15 per cent to 4.2 per cent of the intensity one meter from the source. R_m for minimum audibility was found to be 0.10 ohms.

The following table no. 4 gives a few of the stations observed. Stations numbered 1 to 5 were down the center aisle. The last 4 were on the north side under the balcony. Each determination for any station represents an average of 10 settings for R_n.

TABLE 4.

Station	Distance from source in meters	R_n	I_n	I_n/I_m
0	1	41.1	1.00	170,000
1	10	15.9	.150	25,200
2	18	14.3	.121	20,200
3	28	13.9	.114	19,000
4	37	14.3	.121	20,200
5	45	15.0	.133	22,200
21	28	12.1	.086	14,300
20	36	10.3	.063	10,500
19	43	8.47	.042	7,000
16	52	9.43	.053	8,800

If a sound of constant pitch be long sustained in an auditorium, certain trains of waves will traverse a course which will be little different from that traversed by another set. These will overlap and as a result will either reenforce or mutually destroy each other. This phenomenon is called interference and is common in all types of wave motion. The writer discovered early in the course of this investigation that this

phenomenon existed in the tabernacle and had to be dealt with. Often it was possible by moving the head six inches to move it from a point of maximum intensity to a point of maximum interference. Measurements were made to determine the relation between maximum and minimum intensities at these points. For a point of maximum intensity $R_n = 16$, and for a region of minimum intensity $R_n = 4$, showing a ratio of intensities of 16 to 1. Throughout the experiment an attempt was always made to have the head of the observer in a region of maximum intensity. By slowly walking down the center aisle of the tabernacle the author counted 45 of these regions of maximum intensity through which the head passed.

Articulation Tests in the Salt Lake Tabernacle

In the spring of 1928 this investigation was extended to determine the per cent of, and the variation of, word and syllable reception with a variation of voice intensity, position of auditor in the building, size of audience, etc. The per cent of increase in word reception when electrical speech amplifiers were used was likewise determined. Twelve to fifteen students from the physics department of the Weber College cooperated with the author in making the investigation. The method used was that in which the author called out 75 nonsense syllables or unrelated words in groups of 3. The time required to sound a particular combination of three syllables or words was two seconds and the time between

Table 5.

	Audience	Voice	Per cent reception		
			Minimum	Maximum	Average
Syllable	None	Normal	23	76	39[3]
Syllable	"	3×Nor.	42	76	52[3]
Word	"	Normal	18	68	36[3]
Word	"	3×Nor.	30	84	52[3]
Consonants	"	Normal	42	88	67[3]
Consonants	"	3×Nor.	80	98	93[3]
[1]Words	3100–½ full	3×Nor.	43	70	59[4]
[2]Words	"	Amplifier	64	88	80[4]

[1] 90 per cent maximum single record.
[2] 97 per cent maximum single record.
[3] Average of 3 records in 12 different positions.
[4] 120 records from all parts of the building in 11 different positions.

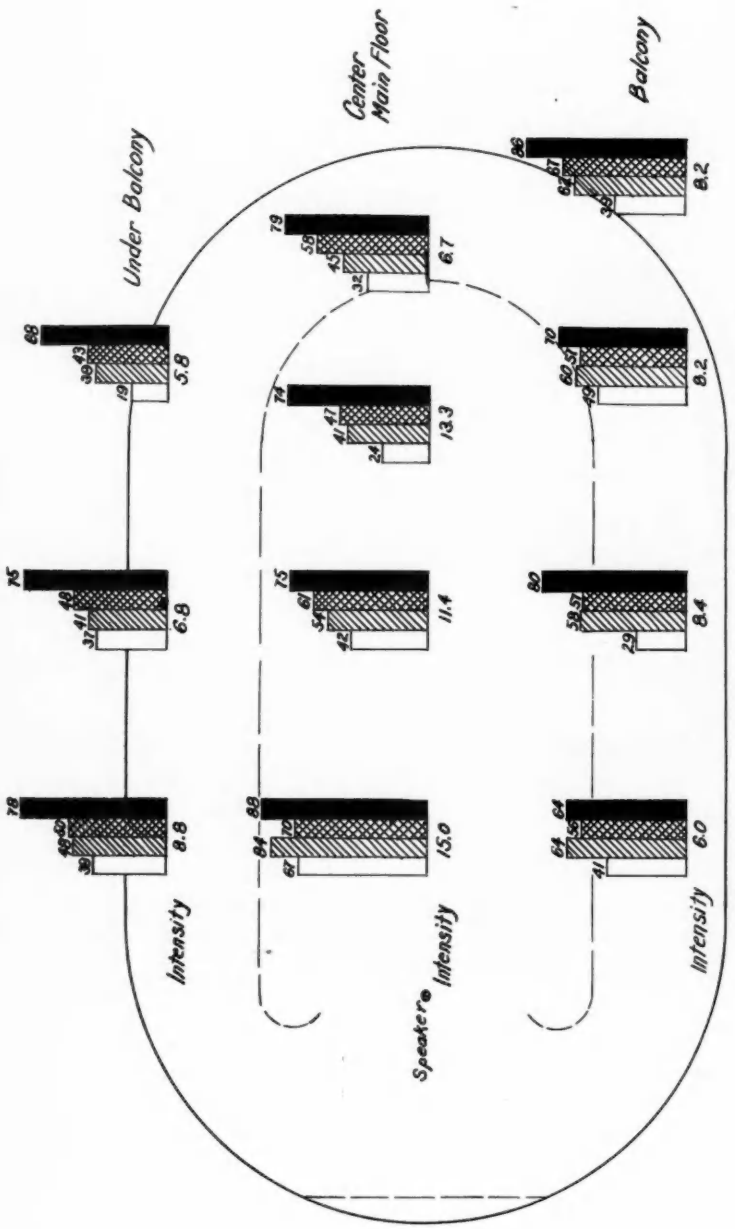


FIG. 3. *Showing variation in word reception with variation in position, voice, intensity, and audience in Salt Lake Tabernacle.*

combinations was approximately $8\frac{1}{2}$ seconds. The auditors (12 to 15 in each position making $12 \times 75 = 900$ syllables heard in each position) wrote what they heard and later compared their results with the originals to determine the per cent of reception. Many preliminary tests were made in various rooms at the Weber College and in the open air to become acquainted with the methods and to standardize the units of loudness.

Two different degrees in the loudness of voice were used—First, that of normal conversation or about 47 sensation units, second, that which was three times normal or that which can be heard three times the distance of normal which would be equivalent to about 57 sensation units.

Results are given in Table 5.

These results are shown graphically in Figure 3.

Conclusions

The conclusions drawn are as follows:

(1) Hearing is obviously best immediately in front of the speaker. It slowly decreases as one recedes down the center aisle and then rises again near the rear.

(2) Hearing is poorest under the balcony and east of the north aisle.

(3) Hearing is better on the whole in the balcony than under the balcony.

(4) Good hearing is fundamentally dependent on loudness of voice increasing from 39 per cent for normal voice to 59 per cent for voice 3× normal and to 80 per cent for the amplified voice.

(5) Good hearing is dependent upon the period of reverberation increasing from 52 to 59 as reverberation decreases from 5 seconds (empty) to 2.3 seconds (3100 persons present).

(6) There is practically no difference in word and syllable articulation in the Salt Lake Tabernacle.

(7) Finally, it is seen that the effects of sound which may exist in a room are loudness or intensity, reverberation, echoes, and interference, the three latter are characteristic of the building itself. It has been shown that the acoustics of any auditorium vary with the number of different factors, namely, volume, absorbing power, audience, pitch, intensity, duration of a sounding source, echoes, interference, etc. In an auditorium the audience may be small, the speaker's voice vary in intensity and pitch and the entertainment a musical number or an

Fig. 3. *Showing variation in word reception with variation in position, voice, intensity, and audience in Salt Lake Tabernacle.*

address, all of which produce widely different effects on a listener. The best arrangement for good acoustics can only be a compromise where the average conditions are fulfilled.

For reverberation this compromise seems to have been well met in the construction of the Salt Lake Tabernacle. When an audience fills this auditorium the reverberation is a little more than a second, a period which is about right to produce the best effects on a listener when the entertainment is musical, and which is a little too long for ideal conditions when the entertainment is an address. It is indeed remarkable for an auditorium of this volume to have a reverberation of less than 5 seconds when the building is empty and about one second when filled with an audience.

The interference by successive reflections which was noted in certain sections of the tabernacle could be greatly reduced by placing some efficient absorbing material on the walls. This would lessen to a great extent the intensity of reflected waves. But since an auditor in the rear of the building (and in the balcony) is so dependent on these reflected waves for good hearing, it is doubtful that anything could be gained.

The author wishes to express his appreciation to Dr. Orin F. Tugman of the University of Utah, Dr. Harvey Fletcher of the Bell Telephone Laboratories, and Joseph Carroll for valuable suggestions and assistance given during the course of this experiment.

Articulation Testing Methods

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This paper is chiefly concerned with the technique of making articulation tests. The construction of a syllabic testing list, the selection of a testing crew, the methods of comparing articulation data for various crews, and the significance of the test as a measure of the speech capabilities of a system are discussed. Various types of lists for different uses are also discussed.

THE transference of thought by means of speech is a very complicated, although common, process. So long as the process runs smoothly, its complications are forgotten. When an auditor fails to understand the speaker, however, inquiry into the reasons for the difficulty begins.

The production, the transmission, and the reception of speech constitute the three important elements of the process. To determine defects in any one of these, it is necessary to have a quantitative means of measuring the recognizability of the speech sounds that the auditor hears. The term "recognizability" as used here refers to correctness with which an auditor identifies a sound as being one, or some combination, of the fundamental speech sounds, when no meaning is associated with the sounds.

During the past few years methods of measuring the recognizability of speech sounds have come into greater and greater use both in this country and abroad. In order to compare the results obtained by various crews in various languages, it is desirable to standardize the methods of test and to set up reference circuits for purposes of calibration. It is the aim of this paper to discuss the methods that have been found the most useful, not only in determining defects in transmission, but defects in the production and reception of speech as well.

One needs only to tabulate the various devices that are used for transmitting speech to realize the importance of a quantitative method of rating their performance. There may be mentioned, for example, the various telephone and radio systems, the phonograph, sound pictures, rooms and auditoriums with various types of acoustic treatment, audiphone sets for the deafened, speaking tubes, etc.

Methods of measuring the recognizability of speech sounds have not been used so extensively for determining the ability of persons to speak properly. Such methods should be of value in the training

of public speakers, actors, students of foreign languages or pupils in deaf schools who are learning to speak.

The rating of auditors by measuring the recognizability of speech sounds which they hear has been used to some extent. For example, such methods have been used to determine the ability of students to interpret a spoken foreign language. Also, the deafness of a person can be determined by such methods. In this case, however, the specialists have usually tried to vary transmission systems between the speaker and listener so as to compensate for the loss of hearing, the amount of such compensation being determined by measuring the recognizability of speech sounds.

The best method of determining the recognizability of the speech sounds naturally depends upon which of the things just enumerated is to be rated. In principle, the method in each case consists in the pronunciation of "selected speech sounds" by a speaker, the transmission of these sounds to an observer's ears, and the recording by the observer of the sounds which he recognizes. Such methods applied to telephone systems have been frequently referred to as articulation tests. The term "articulation" would be more logically used if it were applied only to cases where the speaking abilities of persons are being determined. However, it has been used so frequently in connection with rating transmission systems that it seems convenient to retain it.

The "selected speech sounds" which are ordinarily used in articulation tests are meaningless monosyllables. The percentage of the total number of spoken syllables which are correctly observed is called the syllable articulation. This percentage has frequently been called simply "the articulation."

A syllable is considered to be incorrectly observed, if one or more of the fundamental speech sounds which it contains are mistaken. It is frequently desirable to analyze these mistakes and determine the articulation of the speech sounds. The percentage of the total number of spoken sounds which are correctly observed is called the sound articulation. When the attention is directed toward a specific fundamental sound, such as "b" or "t" or "ā," etc., then the term "individual sound articulation" is used. For example, the individual sound articulation for "b" is the percentage of the number of times that "b" was called that it was observed correctly. Similarly, the terms "consonant articulation" or "vowel articulation" refer to the percentages of the total number of spoken consonant or vowel sounds which are correctly observed.

The articulation values as defined above are taken as the measures

of the recognizabilities of the various phonetic units. English words and short sentences have also been used for testing purposes. When material of this kind is used, a new element enters, namely, the thought or meaning associated with the sentence or word. The criterion for the correct observation of words or sentences is also different from that used in the case of articulation tests. If the thought or meaning of a word or sentence is correctly understood, it is considered to be correctly received, even though the observer may not have correctly recognized each sound that was spoken. The terms "word articulation" or "sentence articulation," therefore, seem inappropriate when referring to the results of such tests. The term "intelligibility" has frequently been used in this sense. Since it has also been used in a more general sense, the terms "discrete word intelligibility" and "discrete sentence intelligibility" will be used when referring to the results obtained by using disconnected words or sentences for the testing material. They are defined as the percentage of the total number of spoken words and sentences, respectively, that are correctly interpreted according to the criterion given above.

Very early in the work of developing the telephone, words and sentences which were chosen in a haphazard way were used for testing purposes. Word lists of various sorts have been worked out and used with some success. Even in very recent years some of these word lists have been used to good advantage. The main objections which have developed, to the continuous use of such lists are: (*a*) it is hard to make the lists equally difficult without resorting to very long lists of words, (*b*) a very large number of lists are necessary in order to avoid memory effects.

Dr. G. A. Campbell [1] was one of the first to propose a system of syllabic speech sounds for testing the transmission characteristics of the telephone system. These syllables had no meaning and were constructed by combining the various initial consonants with the vowel "ee," such as bee, fee, etc. With these lists the consonant articulation was taken as a measure of the system.

Later Dr. I. B. Crandall [2] worked out a system which used both simple and compound consonant forms in a vowel-consonant and consonant-vowel type of syllable. All of the common vowels were used, and the combinations were formed in ways which are usually found in written speech. The sounds occurred with the same frequency as they occur in ordinary written material. As in the Campbell lists, the articulation was based on the consonant sounds alone.

[1] "Telephonic Intelligibility," G. A. Campbell, *Phil. Mag.*, Jan. 1910.
[2] "Composition of Speech," I. B. Crandall, *Phys. Rev.*, 10, p. 74, July 1917.

Several other lists which have not been published were proposed and used, the differences being in the choice of the fundamental speech sounds, in their arrangement into syllables, and in the relative frequency of occurrence, both of the different syllable forms and of the speech sounds in each form. There was a distinct effort to make the lists as nearly like speech as possible by using the syllable forms, and by using the particular combinations of fundamental sounds that occur frequently in English. Difficulties were encountered in testing, however, when this was carried too far in that enough different syllables could not be obtained for continuous testing. On the other hand, when random combinations of sounds were made, without regard to the particular combinations occurring in English, syllables that were very unusual and difficult to pronounce were obtained, unless the combinations were restricted to the simple syllable forms having only two or three sounds. In other words, testing lists must be selected with two things in mind; namely, they must be representative of speech and they must be suitable for making tests. The experience with these various lists also indicated that the results obtained with one system of lists could be calculated approximately from the results obtained with other systems by properly weighting the individual sound articulation values.

This experience led to the adoption by the Laboratories of a system of lists which has been used during the past ten years in studies of the effects of distortion upon the recognition of speech sounds.[3] These lists which have been referred to in the literature as the standard articulation lists were made up of only the con-vow, vow-con and con-vow-con syllable forms. The various fundamental sounds of English were combined at random into the syllables, such that each sound occurred approximately with uniform frequency.

During the past few years it has become evident that still further simplifications in the syllable forms used in the standard articulation lists might be made. Also our methods of making articulation tests and interpreting the results obtained have undergone considerable changes during this time. It is with these new methods that the present paper is chiefly concerned.

In order to distinguish between the old lists and the ones modified as described below, the prefixes "old" and "new" will be placed before the title "Standard Articulation Lists." When there is no chance for confusion, the new lists will be called simply the standard articulation lists, since they are the principal ones now being used in the work at Bell Telephone Laboratories.

[3] "Nature of Speech and Its Interpretation," H. Fletcher, *Journal Franklin Institute*, June, 1922.

New Standard Articulation Lists

In setting up any testing list it is necessary to classify and select a representative group of speech sounds. The National Phonetic Association uses a basic alphabet of 65 different sounds and also uses numerous modifiers which serve to distinguish slight variations in a given sound. Such a system is too complex for testing purposes. The revised scientific alphabet uses 48 simple sounds of which 24 are consonants, 19 vowels, and 5 diphthongs. Besides these fundamental sounds, connected speech contains certain recurrent combinations of them, such as st, ing, etc.

In speech these fundamental sounds are combined into syllables in a large variety of ways, but as mentioned before, in constructing a testing list it is desirable to adhere to very simple syllable forms. More complex forms which include the compound endings are either too few in number or involve unusual speech sound combinations. In either case they are soon memorized by a testing crew working with such lists. In the new lists, therefore, simplifications are made by omitting the con-vow and vow-con types of syllables, leaving only the con-vow-con type. In order to make syllables of this type it is obviously necessary to have the same number of vowels and consonants, provided that each consonant may be used in both the initial and the final position. Some consonants, however, can be used only in the former while others can be used only in the latter position.

With these facts in mind the sounds that are shown in Table I were adopted for these new lists. It will be noticed that all of the consonants are used in both the initial and final positions in the syllable, except h, w, and y, which are used only in the former, and zh, ng, and st, which are used only in the latter position. As was the case in the old standard lists, it will be seen that, in the new lists the vowel variants have been excluded. They occur infrequently in speech and phoneticians do not universely agree on their pronunciation. For this reason they are not included. Also, the diphthongs ī, ou, oi, and ew, which were used in the old lists, were omitted from the new lists. The last two of these diphthong sounds occur very infrequently in speech. Although the diphthongs, ī and ou, do occur quite frequently, it was felt that their essential properties were embraced by the properties of their constituent vowel sounds. By their omission and also by the introduction of the compound st as a final consonant, it is possible to construct any desired number of syllables of the con-vow-con type, from the speech sounds shown in the table.

TABLE I

SPEECH SOUNDS FOR NEW STANDARD TESTING LISTS *

Initial Consonant	I.P.A.	Key Word	Vowel	I.P.A.	Key Word	Final Consonant	I.P.A.
b			a	[ɑː]	father	b	
d			a			d	
f			ā	[eː]	fame	f	
g		go	ā			g	
k			a'	[æ]	fat	k	
l			a'			l	
m			e	[ɛ]	get	m	
n			e			n	
r			ē	[iː]	greet	r	
p			ē			p	
s			i	[ɪ]	tin	s	
sh	[ʃ]	ship	i			sh	
th'	[ð]	this	o	[ʌ]	but	th'	
th	[θ]	thin	o			th	
t			ō	[oː]	go	t	
v			ō			v	
ch	[tʃ]	church	u	[ʊ]	full	ch	
z			u			z	
j	[dʒ]	judge	ū	[uː]	rule	j	
h			ū			zh	[ʒ]
w			o'	[ɔː]	haul	ng	[ŋ]
y	[j]	yawl	o'			st	

Note: Final r and ng are used in the list only when they occur in combination with the following vowels:

a'r (as in carry, paragraph)
ar (as in are, far, tar)
er (as in bury, ferry, verify)
ir (as in spirit)
or (as in her, utter, fir)
o'r (as in for, lore)
ūr (as in your, sure)

a'ng (as in bang, sang)
eng (as in geng, e as in ten)
ing (as in sing, wring)
ong (as in sung, hung)
ung (as in gung, u as in took)
o'ng (as in long, wrong)

* The symbols for the sounds are those of the International Phonetic Association's alphabet. See Pronunciation of Standard English in America, Krapp, Oxford University Press, 1919. See also Revised Scientific Alphabet, Funk and Wagnall's Dictionary.

The testing syllables are formed in the following way. Cards upon which the initial consonants are written are placed in one box; others upon which the vowel sounds are written are placed in a second box; and those upon which the final consonant sounds are written are placed in a third box. A card from each box is drawn at random, thus forming the con-vow-con syllable. By drawing all of the sounds, a list of 22 syllables is formed. This process is repeated three times to obtain a list of 66 syllables which is a unit that has been found convenient to use. A list of syllables of about this length can be used without giving callers and observers a rest period. In such a list each initial consonant occurs three times, each vowel six times, and each final consonant three times.

In forming such syllables only those combinations involving final r and ng that are shown in Table I are included. Much confusion exists concerning the pronunciation of other combinations of these sounds. Syllables that represent slang in English are also omitted. These omissions are made by returning the card upon which the sound in question is written to its box and drawing another card. By combining the sounds at random in this manner any desired number of lists may be made which for practical purposes are all of equal difficulty.

In addition to containing a certain speech sound content, connected speech is characterized by inflection, accent, a rate of utterance, etc. In the earlier articulation studies the test syllables were called singly at intervals of about three seconds. When considered with respect to connected speech this procedure seems somewhat artificial. Comparative tests were made in which the syllables were called singly and as parts of introductory sentences. The tests showed the syllable articulation to be somewhat larger when the introductory sentences were used. The increase was due largely to the greater ease in interpreting the initial consonants of the syllables, when they were inserted in the introductory sentences. The effect was most noticeable for the stop and fricative consonants which have relatively short durations. In order to make the technique more nearly like connected speech the syllables are called in the short introductory sentences. The use of such introductory sentences also helps to insure that any element in the transmission system being tested, whose performance depends particularly upon their immediate past history, will be in the condition in which we are interested for determining speech transmission capabilities.

A list of sentences which is used for this purpose together with a sample record of articulation data is shown in the articulation test record of Table II. For calling purposes, the syllables on the cards are written in the spaces under the columns marked "called" of the test record. These sentences are called uniformly at the rate of 15 per minute. When the syllables in the first column are called, the sentences are repeated using the syllables in the second column and then those in the third column.

The observers are provided with blank articulation test record sheets. They write the sounds which they hear in the corresponding "observed" columns. When the test is completed the observed and called sheets are compared and the various articulations obtained.

For good results it has been found advisable to use a testing crew of ten people—5 men and 5 women. Eight people are ordinarily

TABLE II

Articulation Test Record

DATE 3-16-28 SYLLABLE ARTICULATION 51.5%

TITLE OF TEST Practice Tests CONDITION TESTED 1500~ Low Pass Filter

TEST NO. 10 OBSERVER W.H.S.

LIST NO. 5-9-37 CALLER E.O.

NO.		OBSERVED	CALLED	OBSERVED	CALLED	OBSERVED	CALLED
1	THE FIRST GROUP IS	máv	náv	póz	po'th	Kōb ✓	Kōb
2	CAN YOU HEAR	pōch ✓	pōch	nēz	nēzh	shēth	siz
3	I WILL NOW SAY	seng ✓	seng	jóch ✓	jóch	fūch ✓	fūch
4	AS THE FOURTH WRITE	chūd ✓	chūd	thám ✓	thám	thōl ✓	thōl
5	WRITE DOWN	run ✓	run	hab ✓	hab	poth ✓	poth
6	DID YOU UNDERSTAND	chiz	kiz	def	doth	wám ✓	wám
7	I CONTINUE WITH	foz	fozh	chech	chej	gūm	gūn
8	THESE SOUNDS ARE	lól ✓	lól	lun	lon	nāsh	nāth
9	TRY THE COMBINATION	jās	zhāth	shāl ✓	shāl	vóg ✓	vóg
10	PLEASE RECORD	tháth	thásh	muz ✓	muz	lung	long
11	WRITE THE FOLLOWING	wūr ✓	wūr	led	bēd	diz	dizh
12	NOW TRY	yāp ✓	yāp	wif ✓	wif	Kak	tak
13	THIRTEEN WILL BE	mad	maj	gōst ✓	gōst	thár	zhár
14	YOU SHOULD OBSERVE	bēch	bēk	thav	sāv	must ✓	must
15	WRITE CLEARLY	gēm	dēm	Kōf ✓	Kōf	yód ✓	yód
16	NUMBER 18 IS	theb	veb	rág ✓	rág	jet ✓	jet
17	YOU MAY PERCEIVE	jok	jost	thip ✓	thip	rēp	rēj
18	I AM ABOUT TO SAY	gaf ✓	gaf	yar ✓	yar	thēp	hēp
19	TRY TO HEAR	hus ✓	hus	zhūt ✓	zhūt	——	chuv
20	PLEASE WRITE	hiv	thith	Kūk	tūk	thef	thesh
21	LISTEN CAREFULLY TO	tōg ✓	tōg	fung ✓	fung	bās ✓	bās
22	THE LAST GROUP IS	shót ✓	shót	thev	vesh	thóf	shaf

employed in a test, the remaining two being held for emergencies in order to keep the work going. One member of the crew calls at a time, and the remaining members act as observers. Ordinarily, eight callers are used with four observers recording simultaneously for each caller, although as many as eight or nine observers may be used. The order is arranged such that the various members are equally represented in the test.

Each observer's sheet (Table II) furnishes a value of syllable articulation (the percentage correctly observed), corresponding to a particular caller-observer pair and to 66 syllables or 198 speech sounds called. These values of syllable articulation are recorded in the form shown in Table III. The average of each column gives the average articulation for each observer. The averages of the rows give the callers' articulation.

TABLE III

ARTICULATION TEST RESULT RECORD

FROM 2/29/28 TITLE PRACTICE TESTS
TO 3/21/28 CONDITION 1500 CYCLE LOW PASS FILTER
REFERENCE MM-2186-8/27/28 REMARKS

CONDITION

OBS. / CALLER	E.B.	W.H.S.	F.S.	C.M.	N.C.	A.H.	E.L.F.	M.W.	P.H.		Ave.
E.B.		51.5	51.5	50.0	48.5	57.5	59.0		45.5		51.9
W.H.S.	72.5		62.0	69.5	59.0	64.5	60.5		56.0		63.7
F.S.	47.0	54.5		56.0	48.5	57.5	50.0		42.5		50.9
C.M.	51.5	62.0	53.0		42.5	65.0	57.5		38.0		52.8
N.C.	48.5	56.0	48.5	36.5		52.0	54.5	30.5			46.4
A.H.	41.0	53.0	50.0	53.0	35.0		47.0	42.5			45.9
E.L.F.	51.5	51.5	51.5	46.5	53.0	57.5		41.0			50.2
M.W.	50.0	48.5	50.0	36.5	51.5	59.0	63.5				51.3
Ave.	51.7	53.9	52.4	49.6	48.3	57.0	56.0	38.0	45.5		51.6

When the articulation values are near 100 per cent or 0 per cent, then a group of values will not distribute itself symmetrically about the arithmetic mean or average value. For the high values, this is due to the fact that one cannot get a higher value than 100 per cent. To some observers, the 100 per cent mark may be obtained very easily and to others it may be obtained only with considerable effort. This difference in difficulty cannot be registered in the percentages obtained. A similar reason exists for the unsymmetrical grouping for values near zero. Our experiments have shown that this grouping is symmetrical in the range from 20 per cent to 80 per cent. For the range from 80 per cent to 100 per cent, the average value is less than the most frequent value, and for the range from 0 per cent to 20 per cent, the average value is greater than the most frequent value. These differences are of the order of 1 or 2 per cent. From an extensive series of tests, the averaging factor curve of Fig. 1 was constructed which enables the data to be averaged, in a way, such that the average value is approximately equal to the value which would be most frequently observed in a large number of tests. To do this, each observed articulation, based on 66 syllables, is converted into an averaging factor by means of the above curve. These factors are then averaged and the average value reconverted into average

articulation by means of the above curve. The value so obtained is taken as the average syllable articulation for the test.

The articulations for the various sounds are recorded on the Articulation Test Analysis Record of Table IV. In making the analysis the total errors for each caller are counted. The occurrences per caller are the products of the number of times the sounds are spoken by the caller, and the number of observers. One analysis sheet contains the results for a complete test.

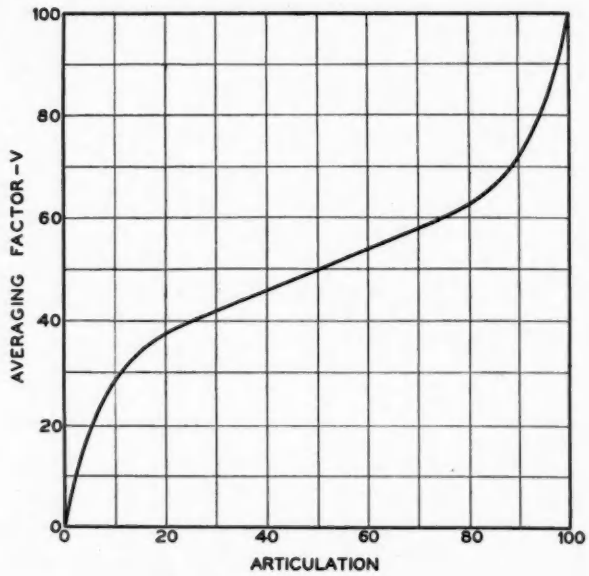

Fig. 1—Averaging factor curve

In dealing with the syllable articulation the unit is 66 called syllables, or the result of one caller and one observer. Hence a number of syllable values are obtained for one test which are averaged by means of the averaging factor curve. When we deal with the articulation of the individual speech sounds, it is advisable to use a larger unit since each sound occurs only six times in a 66 syllable unit. In Table IV the errors are shown for each caller as a unit, and since there were 7 observers per caller, each sound occurs 42 times. This is a unit of sufficient size to qualitatively compare various voices. However, in drawing conclusions as regards the effects of the circuit upon the transmission of the various sounds, it is best to use eight callers as a unit, so that each sound, depending upon the number of observers, occurs of an order of 200 to 300 times. In this case the articulation of an individual sound has a precision that is comparable

with the precision of the syllable articulation when based on 66 called syllables, as will be discussed in a later paragraph.

TABLE IV

ARTICULATION TEST ANALYSIS RECORD

DATE 2/29/28 To 3/21/28 TITLE PRACTICE TESTS

REFERENCE MM-3/26 3/27/28 CONDITION 1500~ LOW PASS FILTER

OBSERVERS EB, WHS, FS, CM, HC, AH, ELF, MN, FH SOUND ARTICULATION 79.3 SYLLABLE ARTICULATION 51.6

LETTER	OCCUR. PER CALLER	ERRORS PER CALLER																		IND. SOUND ART.	
		EB		WHS		FS		CM		HC		AH		ELF		MN		Total Errors	Total Occur.		
a	42	3		2		10		1		15		2		2		3		38	336	88.7	
ā	42	1		1		4		0		7		2		0		2		17	336	94.9	
a'	42	2		0		8		7		20		22		27		11		97	336	71.1	
e	42	2		4		4		4		22		19		20		15		90	336	73.2	
ē	42	0		0		2		0		1		4		0		1		8	336	97.6	
i	42	2		0		2		0		9		8		9		3		33	336	90.2	
o	42	12		1		6		5		12		5		4		10		55	336	83.6	
ō	42	1		0		0		0		0		1		0		0		2	336	99.4	
o'	42	0		0		5		2		0		0		0		0		7	336	97.9	
u	42	5		2		19		1		8		3		4		4		46	336	93.6	
ū	42	1		1		1		1		4		4		2		1		15	336	95.5	
TOTAL	462	29		11		61		21		98		70		68		50		408	3696	89.0	
b	42	7		0		11		10		13		1		4		6		52	336	84.5	
ch	42	13		4		11		13		2		8		8		7		66	336	80.4	
d	42	9		3		6		8		8		7		10		14		65	336	80.7	
f	42	6		11		7		12		25		23		21		22		127	336	62.2	
g	42	9		4		8		13		2		3		1		2		42	336	87.5	
h	21	2		0		0		0		2		0		1		1		6	168	96.4	
j	42	17		4		10		9		3		6		3		2		54	336	83.9	
k	42	17		6		6		11		3		12		11		6		72	336	78.6	
l	42	1		0		3		0		0		2		3		0		9	336	97.3	
m	42	1		2		4		6		2		5		7		0		27	336	92.0	
n	42	4		5		0		10		7		6		9		15		56	336	83.3	
ng	21	0		0		2		3		1		1		4		5		16	168	90.5	
p	42	11		11		4		5		19		21		14		8		93	336	72.3	
r	42	3		0		0		2		4		3		4		2		18	336	94.6	
s	42	18		5		16		9		23		22		18		17		128	336	61.9	
sh	42	25		21		28		26		14		25		17		20		176	336	47.6	
zh	21	6		0		8		2		7		6		2		1		32	168	81.0	
t	42	15		8		15		13		33		24		25		13		146	336	56.5	
th	42	22		24		17		22		27		16		22		27		177	336	47.3	
th	42	20		24		25		19		17		12		20		15		152	336	54.8	
v	42	17		7		12		10		4		9		5		14		78	336	76.8	
w	21	0		0		0		0		0		0		0		0		0	168	100.0	
y	21	1		0		0		0		1		0		1		0		3	168	98.2	
z	21	6		4		11		4		12		9		9		13		68	168	59.5	
zh	42	33		24		32		35		27		27		23		26		227	336	32.4	
TOTAL	924	263		167		236		242		256		248		242		236		1890	7392	74.4	

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SELECTION OF TESTING PERSONNEL

It is necessary to set up the technique of testing, such that the values of syllable articulation can be reproduced within acceptable limits. The limits depend upon the control of the auditory and vocal

characteristics of the testing crew, the control of numerous haphazard factors, and the control of the practice or experience that the crew acquires in the testing of circuits.

The departure from normal, in acuity of hearing of prospective crew members, should be measured with a good audiometer. In our laboratories the 2-A audiometer is used for this purpose. Only those individuals whose average hearing loss departs from normal in the speech range of frequencies (100 to 8,000 cycles per second) by less than 5 db (decibels) are selected. Although of normal hearing, some observers of a crew usually obtain higher values of syllable articulation than do others. The averages of the columns of Table III are a typical set of results for nine observers who have passed such a hearing test and who have had a year or more of experience in observing.

Observer A. H. obtained the highest percentage, namely 59, and observer M. W. obtained the lowest percentage, namely 38. In general this order would be preserved in a series of tests, although haphazard variations in a single test might change the order. The spread in observations is of an order of 20 per cent. More extended tests have shown that the spread tends to decrease as the observed percentages approach 0 or 100. In order to make a replacement in the observing personnel from time to time without causing a probable change of more than 2 per cent in the average percentage, it is necessary to use an observing crew of 8 to 10 persons. Our experience has shown that men and women show no characteristic difference when acting as observers.

The ability of prospective crew members to enunciate the sounds in a normal way is determined in the following manner. An extensive series of tests on various voices have yielded data which are arbitrarily used as a basis for determining normalcy. These tests were made with a simplified list consisting of common English words which will be described in a later paragraph (see Table XVII). Tests were made under three conditions; namely, direct transmission through the air in a quiet, well damped room, transmission over a circuit which uniformly transmitted the frequency range from 100–4,500 cycles, transmission over a circuit having a carbon transmitter. A diagram of this latter circuit is shown in Fig. 7. The sounds were observed by a crew of experienced observers. Table V gives the results of tests that were made upon 21 male and 23 female voices, the personnel being selected from various departments of the Laboratories. The average articulations of the simple consonant sounds are shown. The data are given separately for men and women.

TABLE V

Normal Enunciation

Speech Sound	Air Transmission		Band 100 to 4,500		Carbon Transmitter Circuit	
	Av. % Articulation of Sounds		Av. % Articulation of Sounds		Av. % Articulation of Sounds	
	Men	Women	Men	Women	Men	Women
b	98.7	98.0	96.2	90.1	95.0	91.5
ch	100.0	99.3	98.4	98.0	98.7	98.4
d	99.4	100.0	98.7	98.3	91.9	88.6
f	97.5	84.8	96.5	87.7	79.6	65.2
g	100.0	100.0	99.4	94.6	70.6	63.0
h	100.0	98.0	99.4	98.0	94.4	97.8
k	100.0	100.0	99.1	99.7	78.1	82.1
l	100.0	100.0	98.1	97.7	96.0	87.5
m	99.2	91.3	97.5	95.5	86.5	86.0
n	99.4	100.0	99.4	99.1	95.5	94.5
ng	100.0	100.0	99.7	99.4	93.7	100.0
p	96.9	95.8	97.2	96.6	80.0	73.5
r	100.0	100.0	100.0	99.7	96.9	75.5
s	97.5	99.1	95.0	68.0	91.2	67.8
sh	100.0	100.0	100.0	99.8	98.3	93.1
th	90.4	80.6	75.2	56.3	60.1	50.2
th'	97.5	87.0	93.3	87.7	74.1	77.2
t	98.3	100.0	99.4	96.6	92.9	72.1
v	93.3	78.9	96.2	83.6	87.1	75.0
w	100.0	99.3	100.0	98.9	97.5	84.8
z	100.0	100.0	95.9	70.8	91.2	65.2
Aver	98.2	94.6	96.4	90.2	87.5	80.5

For our work, a prospective crew member is required to call such a list of syllables to a crew of experienced observers. If the observed articulations of the sounds are reasonably close to those indicated in Table V for each circuit, and if no obvious irregularities are noticed in the speech, the prospect is considered satisfactory for testing work. Measurements are also made upon the individual's speech power, but it has not been found necessary to use the information in the process of selection.

Aside from the practical application to the methods of testing, the table is interesting in showing characteristic differences between the voices of men and women. In general, woman's speech is more difficult to interpret than man's, particularly, in the case of the sibilant and fricative consonants. This is probably due to the fact that in woman's speech, these sounds are not only fainter, but occupy higher frequency ranges than in man's speech. The frequency range from 6,000 to 8,000 cycles for the former, is approximately equivalent to the range from 4,000 to 6,000 cycles for the latter. In the case of

the voiced sounds, woman's speech has only one half as many components as man's, which also may cause greater difficulty in interpreting the former.

With respect to the vowel sounds, the crew members are instructed in the correct manner of enunciation. Only those vowels which have definite differences have been included in the testing lists, so that, slight differences in enunciation do not seriously affect the observed results.

The object of the selection process is to determine in a broad but definite way the normalcy in speech of prospective members, and to eliminate those individuals who have speech characteristics which are not readily reproducible should it be necessary to change the testing personnel. The row averages of Table III show a typical set of results for 8 callers who were selected in the above way, and have had a year or more of experience in calling.

The spread in results is of an order of 20 per cent, so that, if a crew of 8 to 10 callers is used, a replacement may be made in the calling personnel without causing a change in the average percentage of more than 2 per cent. Owing to inherent differences in the voices of men and women, they are equally represented on the testing crew. Individuals who have the equivalent of a high school education, and whose ages range from 18 to 23 years, are usually selected for this work.

Control of Haphazard Factors

Haphazard factors arise from various sources, some of which can be controlled reasonably well. The observers work in a sound-proof room, so that extraneous noises will not affect the articulation results. The calling is ordinarily done in a sound-proof booth that has been especially treated with sound absorbing material so as to reduce the reverberation time to an order of a few tenths of a second. Ordinarily the crew does not test more than two to four hours during the day, and the schedule is usually arranged so that this is not done continuously.

The intensity level of each caller is also measured during the test, as small variations in intensity level may cause rather large variations in articulation. Ordinarily the various callers are permitted to call at the intensity level most natural to them, although in some tests the callers all attempt, by watching an indicator, to call at the same level. Various instruments have been used for measuring the intensity levels during tests. The volume indicator [4] has proven quite satis-

[4] This instrument depends for its readings, essentially upon the syllabic powers of the vowel and semi-vowel sounds, so that the reading of the instrument is determined largely by the amplitudes in the frequency range from 100 to 2,000 cycles.

factory and is the instrument ordinarily used for this work. It has the advantage over some of the other instruments that were tried, of being in much more general use on speech circuits.

Control of other haphazard factors of a more or less psychological character, may best be obtained by taking enough data so as to average out their effects. This involves the number of syllables that are called by each speaker and the number of caller-observer pairs that are used in the test. The variability of caller-observer pairs for a calling unit of 66 syllables may be seen from Table III. The probable error [5] in percentage articulation of a single observation (ϵ_s) i.e., one caller-observer pair as taken from the data in the table, is $\pm$ 9. The probable error of the average articulation ($\epsilon_{av.}$) of the 56 caller-observer pairs is $\pm$ 1.2.

It has been found from a large number of tests that the probable error of a number of crews, each consisting of one caller and one observer, is of an order of $\pm$ 12 (per cent articulation) for a 66 syllable unit when the syllable articulation is around 50 per cent. This value tends to decrease with increasing experience in testing, and with increasing or decreasing values of syllable articulation. The use of 36 caller observer pairs obviously reduces the probable error to an order of $\pm$ 2 in percentage articulation, which is about the order of magnitude of the errors involved in maintaining the testing personnel over a period of time.

Since as will be shown in a later paragraph, the syllable articulation is equal to the cube of the sound articulation, the probable error in the sound articulation [6] for one caller and one observer, or a unit of 198 sounds, is of an order of $\pm$ 6 when its value is around 80 per cent. Since each individual sound is called only six times, the probable error for each individual sound for a single caller-observer pair is of an order of $\sqrt{\frac{198}{6}} \times 6 = \pm 35$. If a test comprises 4 observers per caller and 8 callers, each sound is called 192 times, which reduces the probable error for the articulation of each sound to $\pm$ 6. Under

[5] $\varepsilon_s = .67 \frac{\Sigma d^2}{n-1}$ and $\varepsilon_{av.} = \frac{\varepsilon_s}{\sqrt{n}}$; where, n = number of caller-observer pairs; d = difference between the articulation of a caller-observer pair and the average articulation of n caller-observer pairs.

[6] $\varepsilon_s = \frac{\partial S}{\partial L} \varepsilon_L = 3L^2 \varepsilon_L$

$\varepsilon_s = \pm 12$, $S = .5$, $L = S^{1/3}$,

$\varepsilon_L = \frac{\varepsilon_s}{3S^{2/3}} = \pm 6.$

ε_s = prob. error in syl. art. for one caller-observer pair.
ε_L = prob. error in sound art. for one caller-observer pair.

the same circumstances, the probable error for the average syllable articulation is ± 2, and that for the sound articulation is ± 1.

Control of Practice Effects

The third factor entering into the reproducibility of articulation results is practice and experience. The practice effect manifests itself in various ways. An increase in articulation takes place as the observers become more familiar with the vocal characteristics of the speakers. Similar effects are observed as they become more accustomed to a given technique, or to a particular type of distortion. In general, these effects become smaller as the testing crew becomes more experienced.

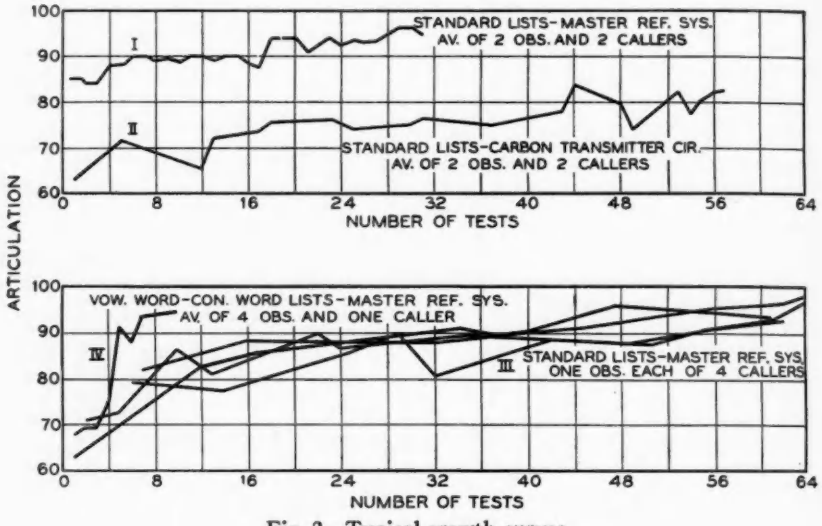


Fig. 2—Typical growth curves

Fig. 2 shows several typical growth curves that were obtained in the process of training new crew members. In this process the new members observe continually on various circuits until the results compare favorably with the results that are obtained by experienced observers. In such tests, experienced speakers are used. The averages for two new observers, of the results that were obtained on a high grade circuit, are shown by Curve I. Two speakers were used in these tests. A limited amount of testing was done by the observers prior to the above tests. Upon the completion of the tests of Curve I about 30 or 40 additional tests were made on various circuits. A

series of tests, in which several speakers were used, were then undertaken on a carbon transmitter circuit. In Curve II the averages for the two observers, of results on two voices, are shown. Three to four weeks' time was spent by the observers in making the various tests mentioned above.

The curves under III show similar data that were taken at a later date by one new observer, for several voices. All of the above tests

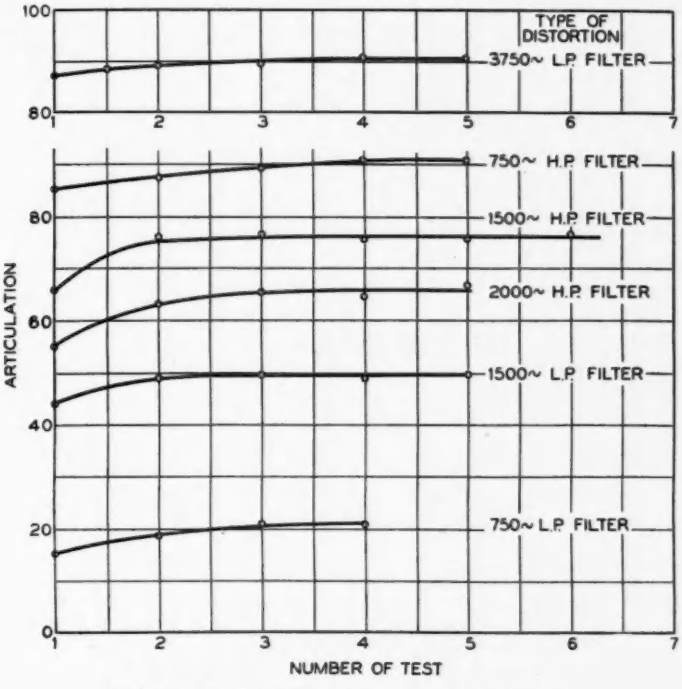


Fig. 3—Practice effects for an experienced crew

were made with the standard lists. In Curve IV, data are shown that were obtained with the vowel and consonant word lists (see Table XVII). In these tests four new observers were used and no preliminary training was given. It is evident that with the word lists, the results reach a state of saturation very quickly.

After a crew has spent several months in testing, its performance becomes largely mechanical. Under such circumstances the practice effects are rather small for types of distorted speech with which it has had experience. When the speech distortion is unusual, however, rather large practice effects may be obtained. Fig. 3 shows such

practice effects for several types of distortion for a crew of eight people. All six of the circuits were tested on each test before going on to the following test. The first three tests were made successively and covered a period of about two months. In each test the types of distortion were interspersed. In other words half of the first test was completed with the filters in one order, and the other half with the filters in the reverse order. The fourth test was made about three

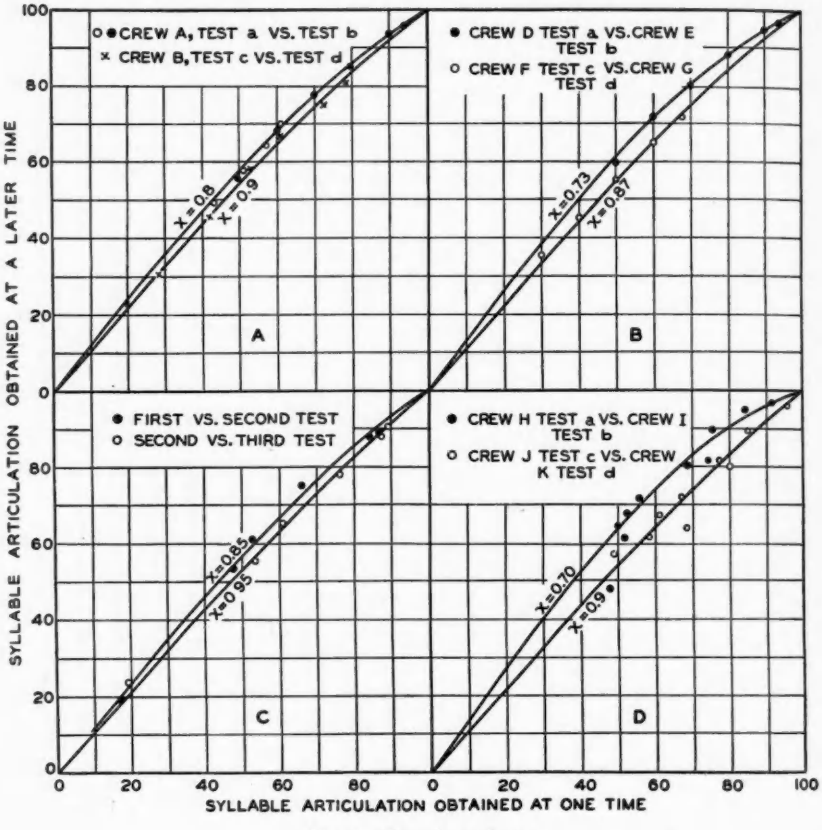


Fig. 4—Practice effects

months later, and the fifth test was made approximately six months later. Although the crew had been testing various circuits for about a year, and were thoroughly accustomed to the routine, these particular circuits had not been previously tested, so that, the crew's experience with these types of speech distortion was small. It is evident that the articulation of an experienced crew reaches saturation very quickly. It is probable that practice effects for a crew of several years' experience with distorted speech would be negligible.

Several procedures are followed in order to correct, in so far as possible, for practice effects. In comparative tests, whenever it is possible, the circuits to be compared are interspersed so as to average out practice effects. If it is desired to compare the articulation of a very new or unusual circuit (from the standpoint of the speech distortion) with one of common experience, several successive tests are made upon the new circuit until no further increase in articulation with practice appears. When it is impossible to intersperse the tests, the data may be corrected to a given state of practice by means of curves which were obtained in the following way. Although as will be seen, this procedure is valid only under certain restrictions, which will be discussed, such a correction will always tend to correct the data to a more comparable basis.

In Fig. 4-*a* a practice curve is shown that was obtained for a crew, from two series of tests that were separated by an interval of three months. The dots represent tests that were made upon a circuit which uniformly transmitted a frequency range from 100 to 5,500 cycles. The circles were obtained from a circuit of the type shown in Fig. 7 involving the carbon transmitter. In both cases the various articulation values correspond to different received speech levels. The crosses represent similar results that were obtained with a different crew on the latter type of circuit.

In Fig. 4-*c* the data of the first three tests in Fig. 3 are shown. In this case the distortion was varied and the received speech level held constant. As previously stated, in so far as was known, the crew had no previous experience with these types of speech distortion so that the practice for the various types of distortion ought to be comparable.

All of the solid curves are graphs of the following equation

$$(1 - S') = (1 - S)^x \cdots, \qquad (1)$$

where S' = decimal value of syllable articulation obtained on a given circuit at one stage of a crew's career,

S = the value obtained on the same circuit at a later stage of the crew's career,

x = a number called the practice factor.

The values of the practice factor x that were necessary in order to fit the observed values are shown in the figure. It is impossible to state definitely that a crew has uniform practice with various types of distortion for the reason that experience is cumulative. A crew's experience with one type of distortion may be of aid in the under-

standing of some other type of distorted speech. With this in mind it will be seen that a constant value of x fits the data for the various types of distortion reasonably well. In the case of changing speech levels with a constant type of speech distortion, where the question of uniformity of experience is not so important, the fit is even better.

It is reasonable to suppose that an inexperienced observer must make a greater mental effort than an experienced observer to obtain the same articulation values. In other words the element reflected

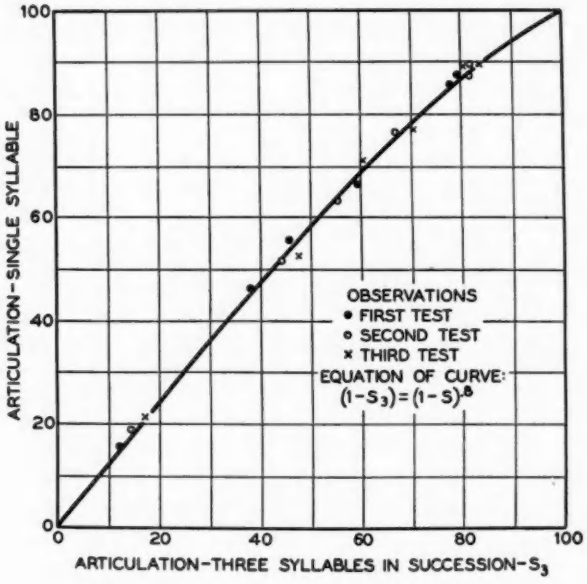


Fig. 5—Relation between techniques

by these curves is closely associated with the burden or strain upon the observer. A somewhat analogous situation obtains when tests are made with two different techniques which differ primarily in the burden imposed on the observer. In making the filter tests described above, two techniques which differed in this respect were used. One was the standard technique, in which one syllable was called with the introductory sentences. In the other, the syllables were called in groups of three (three in succession) with the sentences. The syllables were uttered as nearly in the manner of a three syllable word as was possible.

The results that were obtained with the two techniques are shown in Fig. 5. When the syllables are called in groups of three, the articulation values are smaller than when they are called singly.

It was found, however, that the type of relation shown in Eq. 1 also relates the data obtained with the two techniques. In this case the relation may be expressed as follows:

$$(1 - S_3) = (1 - S)^{\cdot 8} \cdots, \qquad (2)$$

where S_3 = decimal value of syllable articulation when called connectedly,

S = decimal value of syllable articulation when called singly.

The curve of Fig. 5 is a graph of the above equation.

In this case uniformity of experience with the various types of distortion does not enter, as the tests with the two techniques were made simultaneously. The only difference in the techniques was that in the three-syllable case the observer listened to three syllables before writing them down. It seems reasonable to conclude, therefore, that when a crew has the same experience with different types of distortion, then the results obtained by it at one time may be compared with the results obtained by it at some other time by using such a relation. No doubt other types of functions could be found which would also fit the above data. The relation shown here was chosen because it fit both the practice data and the data that were obtained with the two different techniques and is very convenient to use in making such corrections.

It is evident that in order to use the practice curves it is necessary to set up a reference circuit in order to obtain an appropriate value of x. Theoretically, one reference condition should be sufficient, provided that the practice of the crew had the same relative distribution over various types of speech distortion. Since this is usually not the case, it is necessary to use several reference circuits representing various types of speech distortion. When it is desired to correct data for practice effects, the appropriate value of x is determined by making tests upon the reference circuits having types of distortion similar to the circuits for which the corrections are desired. A description of several reference or control circuits which have been found useful with the values of sound and syllable articulation as obtained with the testing crew of five men and five women as previously described, is given below.

(*a*) *Air Transmission. Master Reference System for Telephone Transmission.*—The air transmission tests were made in a quiet, well damped room having a volume of approximately 1,000 cubic feet. The observers faced away and were located at an average distance of 30 inches from the speaker. Sound articulation "L" 99.1 per cent.

Syllable articulation "S" 97.5 per cent. Practically identical results were obtained with the "Master Reference System" [7] with the system set for optimum received speech level, i.e. a sensation level of 70 db, average distance from lips to transmitter 1.5 inches.

(*b*) *Auxiliary Circuit of the Master Reference System.*—The auxiliary circuit of the master reference system consists of networks which are

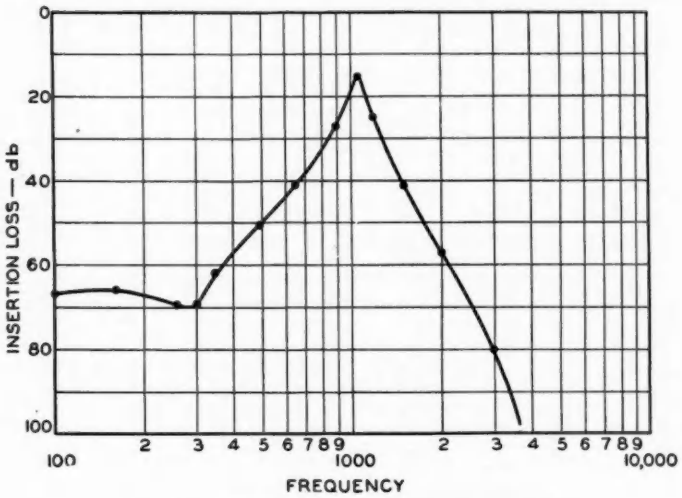


Fig. 6—Insertion loss of auxiliary networks

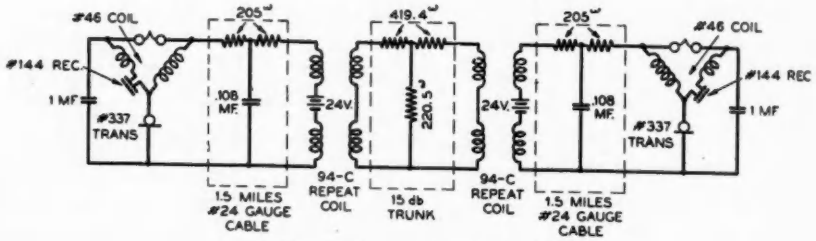


Fig. 7—Carbon transmitter circuit

inserted into the otherwise distortionless reference system, to give it a frequency resonance around 1,100 cycles. The insertion loss of the networks is shown in Fig. 6. This loss is approximately equal to the combined losses of the No. 1 transmitter and receiver distortion networks of the Master reference system. Sensation level 74 db $L = 89.2$ per cent, $S = 72$ per cent.

[7] L. J. Sivian, "A Telephone Transmission Reference System," *Electrical Communication*, 3, Oct. 1924. M. Cohen, "Apparatus Standards of Telephonic Transmission." W. H. Martin and C. H. G. Gray, "Master Reference System," *Bell Tech. Jour.*, July, 1929.

(*c*) *Carbon Transmitter Circuit* (see Fig. 7).—The average values for five transmitters are $L = 93$ per cent, $S = 81$ per cent. In these tests the sensation level of the received speech was 75 db, and the calling level as measured by a volume indicator bridged across the line side of the input repeating coil was − 12.5 db.

(*d*) *Master Reference System Plus Filters.*—System set for a sensation level of 70 db.

3,750~ Low Pass Filter	$L = 96.7\%$	$S = 91.0\%$
750~ High " "	$L = 96.7\%$	$S = 91.0\%$
1,500~ Low " "	$L = 77.7\%$	$S = 49.5\%$
1,500~ High " "	$L = 91.0\%$	$S = 76.0\%$

The foregoing discussion has been concerned with methods of correcting the articulation results obtained by a given crew at different times to an arbitrary stage of practice or experience. To do this it is necessary to calibrate the crew for types of distortion that are similar to those of the systems for which the corrections are desired. The method has been described in detail because there are times when it is necessary to make such corrections. However, it has been our experience that such practice effects become negligible with a crew that has been set up in accordance with the methods previously described, when the crew's experience with types of distortion is diversified and when unusual circuits are tested successively until no further increase in articulation with practice occurs.

These methods may also be used to correlate the articulation data of various crews and various techniques, provided that the only essential difference between the crews and techniques is in the demand or burden that is placed upon the observer. This means that the crews must have similar vocal characteristics and similar hearing abilities, and that the testing lists must have similar speech sound content. It has been found, for example, that a crew of women callers obtain a considerably higher articulation than men callers on a circuit which eliminates all frequencies below 1,500 cycles and vice versa on a circuit which eliminates all frequencies above 1,500 cycles. It is obvious, therefore, that the methods described above could not be used to correlate the two crews for such circuits. Similarly, the methods could not be used for comparing two crews, if the hearing level of one is 10 db below the other, or to compare two techniques, one of which is made up entirely of vowel sounds and the other entirely of consonant sounds. As shown in Fig. 4-*b*, data have been obtained with various crews on various circuits which can be correlated very

well by means of the above curves. Fig. 4-*d* gives data that were obtained with other crews which show very poor correlation. In neither case are the characteristics of the crew well enough known to satisfactorily account for the observed differences. At the time the work was done the significance of these factors was not so apparent, so that they were not given the attention they now receive. During the past two years a crew of 10 people has been used almost continuously in testing work. During this time numerous changes in personnel have taken place and only five of the original members are now on the crew. The data obtained during this time appear to be strictly comparable. In some cases it is necessary to use the practice curves. In other cases (circuits that are frequently tested), practically identical results are obtained. For this reason, it is believed that if a similar crew of 10 different people were to be selected as previously described, comparable articulation results would be obtained. It seems reasonable to expect that crews testing in various languages should also obtain comparable results provided that the crews were similar in the sense used here and that the lists were phonetically similar. It seems desirable, therefore, to standardize on the factors which affect the comparison of data, such as, the size and type of crew, the type of list, and the type and number of reference circuits. Best results are likely to be obtained when the crews do not differ by amounts which correspond to values of x less than 0.7. Smaller values indicate that the crews have not had sufficient testing experience, or have speech and hearing characteristics which are essentially different, or that the phonetic content of the testing lists are appreciably different. In the latter case the results may be correlated by means of statistical relations that will be given in a later paragraph.

Relation of Articulation to the Transference of Thought by Speech

The foregoing paragraphs have been concerned with the practical problems of setting up a suitable testing technique and correlating the observed articulation results. The procedure that has been discussed enables us to measure the percentages of the various speech sounds which are correctly recognized when they are spoken in a simple con-vow-con syllable. We desire at this point to consider the broader significance of this measure. In other words, how is the articulation result related to the transference of thought by means of speech? This relationship involves many psychological factors which are difficult to evaluate so it must not be expected that a comprehensive answer can be given here, but it is important to understand

as fully as possible those parts of the problem that can be evaluated. Such a relation involves two questions, (*a*) how do the articulations of the sounds as measured with the testing lists compare with their articulations as they are used in speech, (*b*) how should the articulation values be weighted in order to obtain an index of the speech capabilities of a system.

In the first place, certain fundamental sounds of speech were omitted from the above lists. The most important of these are the consonant compounds. The majority of these sounds may be regarded as the product of a very few combining consonants acting as modifiers to the rest of the consonant alphabet. Since the combining consonants or modifiers occur over and over in combination with various consonants, it might be expected that the interpretation of the compounds would depend primarily upon the interpretation of the various consonants, and not upon the modifiers. In other words, the compounds would be interpreted as simple consonant sounds. The tests discussed below show that this is true on the average, although notable exceptions may occur in individual cases.

The testing lists were made up from the sounds shown in Table VI.

TABLE VI

Consonants		*Vowels*
Initial	*Final*	
b, br,	rb, b,	a'
d, dr,	rd, d,	a
g, gr,	rg, g,	e
p, pr,	rp, p,	i
k, kr,	rk, k,	o
t, tr,	rt, t,	
f, fr,	rf, f,	
th, thr,	rth, th,	
s, sl,	nd, d,	
b, bl,	nj, j,	
g, gl,	nz, z,	
p, pl,	nk, k,	
k, kl,	nt, t,	
f, fl,	ns, s,	
r, l,	n, r,	

These sounds were combined at random into syllables of the con-vow-con-con and con-con-vow-con form. Ten lists of 90 syllables each were made, and a crew of 10 callers, with 5 observers per caller, was used. With this number of tests the probable error in the per cent articulation for each sound is approximately 5 per cent. The tests were made on the auxiliary circuit of the master reference system. The sensation level of the received speech was about 80 db. The results are shown in Table VII.

TABLE VII

ARTICULATION OF CONSONANT COMPOUNDS

Initial				Final			
Sound	% Art.	Sound	% Art.	Sound	% Art.	Sound	% Art.
b	89.0	br	85.3	rb	63.3	b	77.0
d	98.0	dr	88.7	rd	60.0	d	91.0
g	96.3	gr	90.0	rg	57.3	g	88.7
p	78.3	pr	52.0	rp	39.3	p	66.7
k	95.3	kr	93.3	rk	52.7	k	90.7
t	79.3	tr	89.3	rt	58.7	t	88.7
f	56.3	fr	60.7	rf	39.3	f	53.3
th	71.3	thr	88.0	rth	42.0	th	52.0
ave.	83.0	ave.	81.0	ave.	51.5	ave.	76.0
s	57.3	sl	72.7	nd	92.7	d	91.0
b	89.0	bl	95.3	nj	91.3	j	96.0
g	96.3	gl	77.3	nz	82.0	z	76.7
p	78.3	pl	68.0	nk	92.7	k	90.7
k	95.3	kl	86.7	nt	84.7	t	88.7
f	56.3	fl	86.7	ns	72.7	s	44.7
ave.	78.7	ave.	81.1	ave.	86.0	ave.	81.3
Init. Ave.	81.1		81.0	Fin. Ave.	66.3		78.3

Ave. Simple Cons. 79.7
" Comp. " ..... 73.5 Exclusive of r () 82.7

The articulation of the consonant compounds as a class does not differ appreciably from the articulation of the corresponding simple consonants. The final r compound is seen to be an exception to this general rule. The errors for combinations containing this sound were caused by the large number of omissions of the modifier. For example, if "barb" were called, "bab" would be recorded. When the final r is combined with a consonant, the tendency is to shorten its duration and to stress it less than is done when it occurs as a simple final. Also, as mentioned before, the r sound materially modifies the vowel preceding it and usually in such a way that the vowel and r sounds are spoken as a vowel. For these two reasons, it escapes detection more readily than when used as a simple final. It will be noticed from the table that f is definitely more difficult to recognize than fl, while p is definitely less difficult to recognize than its compounds pl and pr. There is also a large difference between the results for s and ns. Although these differences are large, some tend to increase and others to decrease the average articulation. It is seen from the table that if the r compounds are omitted, the averages for the simple consonant sounds and for their compounds are approximately equal. Since this class of sounds comprises less than 15 per cent of the speech sounds, the results obtained by using a list in which the consonant

compounds are omitted will be very closely the same as those obtained by lists in which such sounds occur. In view of this, and also because their inclusion would greatly extend the time needed for testing, compound consonants have been omitted.

In conversational or written speech some of the sounds are used much more frequently than others, whereas in the testing lists each sound is used the same number of times. Does this procedure lead to essentially different articulation values, for the various sounds, from those obtained by using the sounds in proportion to their frequencies of occurrence in speech?

TABLE VIII

ARTICULATION OF SOUNDS OF EQUAL VS. UNEQUAL OCCURRENCE

Sound	Equal Occ.		Unequal Occ.	
	No. of Occur.	Art.	No. of Occur.	Art.
a	300	91.0	550	88.0
ā	300	98.3	600	96.5
a'	300	88.3	400	82.3
e	300	91.0	1200	87.4
ē	300	100.0	850	98.8
i	300	96.0	1150	94.5
o	300	86.0	500	90.6
ō	300	98.3	500	97.6
o'	300	97.7	450	96.2
u	300	88.3	150	92.7
ū	300	97.3	250	97.6
Ave.		93.8		92.9
b	300	87.0	400	91.0
ch	300	97.3	150	98.0
d	300	91.0	700	89.1
f	300	73.7	400	67.8
g	300	92.0	450	94.2
h	150	100.0	200	100.0
j	300	98.0	150	97.3
k	300	87.3	900	89.8
l	300	98.7	1200	96.2
m	300	95.3	900	81.3
n	300	93.7	1300	95.5
ng	150	93.3	400	90.0
p	300	75.3	500	71.2
r	300	97.0	1250	97.4
s	300	72.3	850	76.3
sh	300	99.3	300	99.7
st	150	96.0	200	96.5
t	300	83.7	1550	86.6
th	300	60.3	200	60.5
th'	300	70.7	150	68.0
v	300	78.7	250	71.6
w	150	96.7	400	97.8
y	150	99.3	100	94.0
z	300	67.0	250	72.0
zh	150	97.3	50	100.0
Ave.		88.0		87.3

Table VIII gives the results of articulation tests that were made with two such types of lists. In both cases the sounds were combined at random into syllables of the con-vow-con type. The tests were made on the auxiliary circuit of the master reference system.

Realizing that the probable error in the articulation value given for each sound is ± 5, there do not appear to be any outstanding differences in the articulations of the various sounds with the two types of list. The average articulations for the two lists differ by less than the probable error. The test indicate, therefore, that lists having uniform occurrence of sounds give the same individual sound articulation values as lists having the frequencies of occurrence of the sounds proportional to their frequencies of occurrence in speech. At least this is true within the accuracy usually attained in making such tests. The testing advantages of the former type of list have already been pointed out.

It is important to notice that the average sound or the average syllable articulation may not be the same for the two types of lists. even though the articulation for each sound is the same. The averages shown in the table were obtained by assigning equal weights to the articulation for each fundamental sound. If weights which are proportional to frequency of occurrence of the sounds in speech be assigned, the averages obtained will, in general, be slightly different. For the particular circuit corresponding to the data of Table VIII, the averages obtained in the two ways did not differ by more than the observational error. Our data have shown that this is also true for a large class of circuits ordinarily used in telephone work. However, those transmission systems which have a specific effect upon certain consonant or vowel sounds, for example, upon s which occurs 850 times in one list compared to 300 times in the other, would obviously have different values for the sound articulation by using the two methods of obtaining the average.

In speech, certain combinations of sounds occur more frequently than others. In other words, some consonants precede certain vowels more frequently than they do other vowels, and similarly, some consonants follow certain vowels more frequently than others. For example, the combination "es" is used much more frequently than the combination "us" (u as in foot). Since the testing lists are made by random selection, the various con-vow and vow-con combinations occur with uniform frequency. In order to determine how this difference influences the interpretation of the sounds, articulation data on various circuits were examined. Attention was focused first on the final consonant sounds. One hundred errors for each consonant,

or 2,200 consonant errors were selected at random from the articulation data and the number of these 2,200 errors that occurred after each vowel sound ascertained. Similarly, the number of vowel errors, out of a total of 1,100 errors, that occurred after each of the consonant sounds, was determined.

Probability studies indicate that the distribution of these errors as shown in Table IX is of the same order as that to be expected on

TABLE IX

DISTRIBUTION OF VOWEL AND CONSONANT ERRORS

Distribution of Vow. Errors		Distribution of Fin. Con. Errors	
Preceding Con. Sound	No. of Vow. Errors	Preceding Vow. Sound	No. of Fin. Con. Errors
b	58	a	188
ch	73	ā	264
d	52	a'	232
f	49	e	166
g	50	ē	214
h	55	i	172
j	54	o	192
k	60	ō	192
l	45	o'	208
m	49	u	196
n	54	ū	176
p	68		
r	39		
s	50		
sh	53		
th	38		
t	52		
v	40		
w	52		
y	49		
z	60		

the basis that the distribution of errors is due entirely to chance. Since, from the way the lists were constructed, the occurrence distribution is due to chance, it is evident that the errors in recognition of the sounds do not depend upon the particular sounds that they follow. Although the analysis was not made, it would be expected that a similar situation obtains for initial consonant and vowel errors. These data may be interpreted to mean that the consonant articulation, the vowel articulation, the sound articulation, or the syllable articulation, is approximately independent of the particular sound combinations, when a wide variety of combinations are used. The results obtained with these lists, therefore, are as representative of speech as the results that would be obtained with lists employing particular sound combinations in proportion to their frequencies of occurrence

in speech. The analysis was not extensive enough to draw conclusions as to the effects of particular sound combinations upon the articulation of individual speech sounds.

Approximately 40 per cent of the syllables that occur in English are of the con-vow-con type. About 34 per cent are of the con-vow, and vow-con type. The syllables, including the compounds, such as, con-con-vow, vow-con-con, con-vow-con-con, and con-con-vow-con, make up about 16 per cent of the syllables of English. Since, as pointed out above, the interpretation of the consonant compounds depends primarily upon only one of the consonants, the latter syllables may be grouped in the two former classes, which then constitute some 90 per cent of English. Of the remaining syllables, 7 per cent consist of a single vowel, so that the more complex syllable forms constitute only 3 per cent of English.[8] Since 97 per cent of the syllables of English are included in the one, two and three letter forms, there is little reason to include the more complex syllable forms in order to represent speech, when as has been previously stated, they are undesirable from a testing standpoint. As will be shown in a later paragraph, one, two and three-letter syllables all yield equal values of articulation for the various speech sounds. Since the three-letter syllables require a smaller testing time for a given number of called sounds, the other syllable forms were excluded from the testing lists.

Having shown that the standard technique gives, for the various sounds, data that are representative of speech, the question now arises as to the best figure that may be computed from the data obtained with this technique, in order to best represent the speech transmission ability of the system under test. Before discussing this, it is necessary to consider some probability relations existing between the quantities entering into the calculation of such a figure.

Statistical Relations

The syllable articulation S when expressed as the ratio of the number of successes (correct interpretations of the syllables) to the number of trials (syllables called) is the chance of perceiving a syllable correctly. Also, if a similar ratio is used for the sound articulation L, the vowel articulation V, and the consonant articulation C, then these letters represent the probability of perceiving correctly a fundamental sound, a vowel sound or a consonent sound, respectively. If a syllable contains only one fundamental sound, then it is obvious that

$$S = L. \tag{3}$$

[8] These data were obtained from Godfrey Dewey's book "Relative Frequency of English Speech Sounds," Harvard University Press, 1923.

If a syllable has two letter sounds, then the chance of perceiving them both correctly is the same as the chance of perceiving the syllable correctly or

$$S = L^2. \tag{4}$$

Similarly, for a syllable containing m sounds

$$S = L^m. \tag{5}$$

Or if A_1, A_2, A_3, $A_4 \cdots A_m$ give the per cent of syllables in the list containing 1, 2, 3, 4, $\cdots$ m sounds, respectively

$$S = A_1L + A_2L^2 + A_3L^3 + \cdots A_mL^m. \tag{6}$$

Similarly, the chance of perceiving a syllable of the type con-vow or vow-con is VC; of the type con-vow-con, con-con-vow or vow-con-con is VC^2; of the type con-con-vow-con, con-vow-con-con, vow-con-con-con, or con-con-con-vow, is VC^3, etc.

For the old standard articulation lists these formulæ reduced to

$$S = \frac{1}{5}VC + \frac{4}{5}VC^2 = \frac{1}{5}L^2 + \frac{4}{5}L^3. \tag{7}$$

For the new standard articulation lists they reduce to [9]

$$S = VC^2 = L^3. \tag{8}$$

If a list of N syllables is used, then the letter errors and syllable errors will be $3N(1 - L)$ and $N(1 - L^3)$, respectively, or the number of letter errors per mistaken syllable, for the new standard lists, will be

$$m = \frac{3}{1 + L + L^2}. \tag{9}$$

It is seen that m approaches 3 as L becomes small, and unity as L approaches unity. For $L = .30$, $m = 2.06$; for $L = .50$, $m = 1.71$;

[9] When derived from the probability formulæ

$$VC^2 = L^3.$$

However, from the definition of V, C and L,

$$L = (2C + V)/3 \quad \text{so that} \quad VC^2 = L^3.$$

The difference is

$$d = \frac{(V + 8C)(V - C)^2}{27}.$$

Actually V and C are not wholly independent of each other and when values as obtained in tests are substituted in the above equation, the difference turns out to be small.

and for $L = .80$, $m = 1.23$. When observed values of m become consistently greater or less than this theoretical value, it must be concluded that the assumptions underlying this statistical theory are not valid.

All of the above statistical relations are dependent upon the tacit assumption that the chance of perceiving any sound correctly is entirely independent of the other sounds present and also independent of the number of other sounds present. It was shown in the previous section that the articulation of the various sounds is, on the average, independent of the other sounds in the syllables. On the other hand, experiments have indicated that the articulation does depend upon the number of sounds in the syllable. The sound articulation becomes smaller when the number of sounds in the syllables increases beyond three per syllable.

The data from which this conclusion was drawn were taken from three different experiments. In the first, three different transmission systems were tested by using first the standard articulation lists and then the vowel-consonant lists which are described in the last section. When using the vowel list, the vowels only are considered and when using the consonant list the consonants only are considered. These lists together, then may be considered as composed of syllables having only one sound. The syllable and sound articulations are the same when using such lists. The comparison of the results obtained with the two types of lists is shown in Table X. It will be seen that there

TABLE X

ARTICULATION FOR ONE- AND THREE-SOUND SYLLABLES

	Freq. below 1,000~ only			Freq. below 1,950~ only			Freq. above 1,500~ only		
	Vow. Art.	Cons. Art.	Sound Art.	Vow. Art.	Cons. Art.	Sound Art.	Vow. Art.	Cons. Art.	Sound Art.
One-sound Syllables	71.5	62	65	98.5	82.5	88	81	96	91
Three-sound Syllables.....	69.5	61.5	64	96	83	87	80	96.5	91

is only a slight tendency for the sound articulation to be lower for the three-sound syllable when compared with the one-sound syllable. The differences are within the observational error in testing.

In the second experiment a new list was constructed using the syllable forms con-vow and vow-con. The auxiliary circuit of the master reference system was tested with this list and also with the standard articulation list. The results are shown in Table XI. The

TABLE XI

ARTICULATIONS FOR TWO- AND THREE-SOUND SYLLABLES

	V	*C*	*L*
Two-sound Syllables	95	88	90
Three-sound Syllables	94	87	89

number of syllables used of each type for determining these averages was 1,344. The average sound articulation in each case was determined by giving equal weights to the articulation for each sound. For the three-sound syllables this is done by dividing the number of sounds correctly recognized by the total number of sounds called. For the two-sound syllable the procedure is not so simple. Since each vowel sound occurs twice as often as each consonant sound, it is necessary to obtain an average for the vowels and consonants

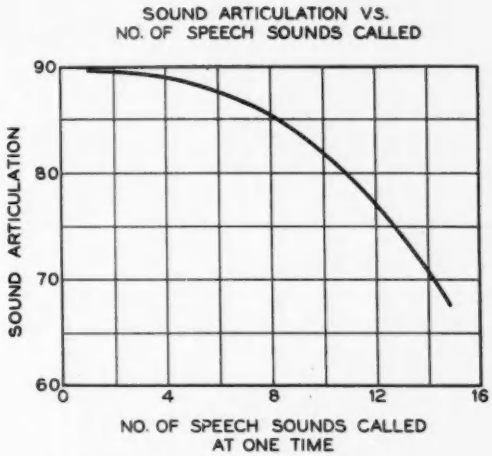


Fig. 8

separately. The final average value for the sound articulation is obtained by assigning weights of 1 and 2 to the vowel and consonant articulations, respectively. It is seen that there is no appreciable difference between the values of *L* obtained by the two types of lists.

In the third experiment the standard lists were used to test the auxiliary circuit but the syllables were called in groups of 1, 2, 3, 4, or 5 at a time. The results of these tests are shown in Table XII.

TABLE XII

Number of Sounds Called at One Time	Sound Articulation
3	89.0
6	87.5
9	84.0
12	77.0
15	67.0

From these three sets of data and from other available data which could be applied to this problem, the curve shown in Fig. 8 was constructed. It gives the sound articulation which would be obtained for a circuit such as the auxiliary circuit of the master reference system, when the number of sounds that are spoken at a time, that is, before the observer starts writing, is represented by the abscissa. It is evident from the shape of this curve that the assumptions underlying the statistical formulæ are valid for syllables having three or less sounds per syllable, and that they will break down for the more complex types of syllables. These assumptions might be expected to break down also, for certain extreme types of distortion.

Definite relations between the vowel, consonant, sound, and syllable articulations for both the old and the new techniques, have been derived by statistical theory. An experimental relationship between these quantities is shown in Figs. 9 and 10. These were obtained by an analysis of the errors of a large number of tests with widely different types of distortion, the data in Fig. 9 being taken with the old and the data in Fig. 10 with the new technique.

In the figures observed values of sound articulation have been plotted against the corresponding observed syllable articulation values. The solid curves in the two cases were calculated from Equations 7 and 8, respectively. The observed values agree reasonably well with the theoretical curves.

There is very little correspondence between the vowel and syllable or consonant and syllable articulation. The table below shows that

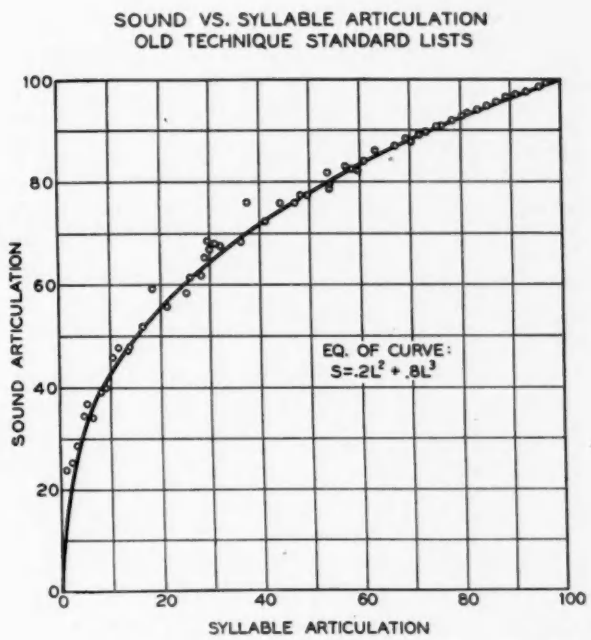
TABLE XIII

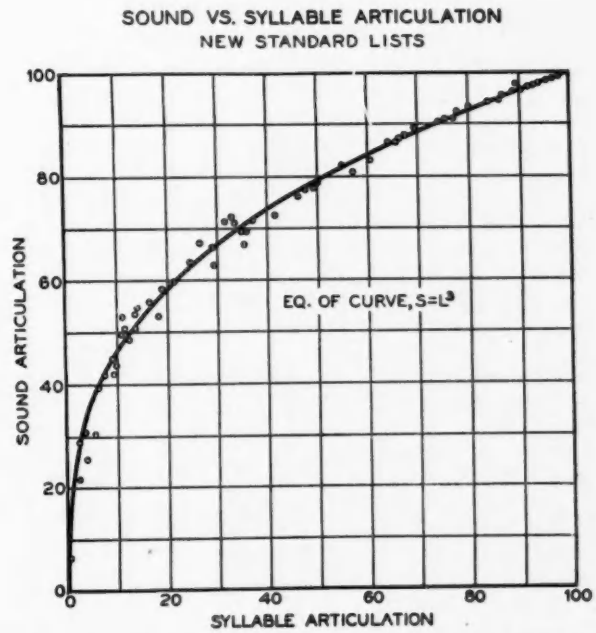
VOWEL, CONSONANT, AND SYLLABLE ARTICULATIONS

	V	C	S	VC^2
3,750 L.P.F.	98.7	95.4	90.8	89.8
750 H.P.F.	93.0	98.6	90.9	90.4
2,850 L.P.F.	98.5	92.9	86.3	85.4
1,000 H.P.F.	89.0	98.4	87.0	86.2

a circuit which discriminates against the vowels may have a syllable articulation equal to another circuit which discriminates against the consonants. However, it is seen that the product VC^2 is equal to S as the statistical theory indicates. If then *the sound, or vowel and consonant articulations are known, it is possible to calculate the syllable articulation,* for the case of two- and three-sound syllables.

We are now in a position to consider the figure which best represents





Figs. 9 and 10—Relation between sound and syllable articulation

the capabilities of systems to transfer thought by means of speech. For giving a complete picture, it is necessary to give the articulation values for each speech sound. Since this involves 36 articulation values, it is difficult to compare various systems. To combine these values into averages raises the question of how such an average shall be taken. At first thought it might seem obvious that the weights assigned to each sound articulation value should be proportional to the frequency of occurrence of that sound in English speech. Many of the most frequently occurring words, however, such as the, of, and, to, in, that, etc., do not carry much of the thought, so that it seems reasonable to exclude the effects of such words in the weighting process. It is evident that many sets of weighting factors could be evolved depending upon how far the exclusion process is carried and depending upon whether written or spoken English is used, in determining the frequencies of occurrence of the sounds. After excluding the twenty or twenty-five most common words, however, further exclusion does not appreciably change the calculated articulation value. The table below gives a set of factors obtained from the frequencies of occurrence used in Table VIII. They are based upon the studies of Messrs. French and Koenig [10] on the frequencies of occurrence of speech sounds in spoken English. The effects of the more common parts of speech, such as, personal pronouns, definite articles, conjunctions, and prepositions have been excluded.

TABLE XIV

Group I	Weight	Group II	Weight	Group III	Weight	Group IV	Weight	Group V	Weight
ā	3.0	i	5.8	r	6.3	d	3.5	z	1.3
ē	4.3	o	2.5	l	6.1	t	7.8	s	4.3
ō	2.5	a'	2.0	ng	2.0	b	2.0	v	1.3
ū	1.3	u	.8	n	6.6	p	2.5	f	2.0
o'	2.3	e	6.0	m	4.5	g	2.3	zh	.3
a	2.8	y	.5			k	4.5	sh	1.5
		w	2.0			j	.8	th'	.8
						ch	.8	th	1.0
						h	1.0	st	1.0
Total Weight	16.2		19.6		25.5		25.2		13.5

It will be noticed that the speech sounds are arranged in five groups. The sounds in each group have very similar characteristics, so instead of dealing with 36 articulation values for a circuit, it is only necessary

[10] "Frequency of Occurrence of Speech Sounds in Spoken English," N. R. French & W. Koenig, *Proc. Acoustical Society of America*, 1929.

to deal with the average value for each of the five groups. The average for the first group is designated V_l, signifying long-vowel index; for the second group V_s, signifying short-vowel index; for the third group C_n, signifying nasalized-consonant index; for the fourth group C_s, signifying stop-consonant index; for the fifth group C_f, signifying fricative-consonant index. If the articulation obtained from any test for each sound be designated by the phonetic symbol for that sound, then,

$$\left.\begin{aligned} V_l &= .19\ \bar{a} + .27\ \bar{e} + .15\ \bar{o} + .08\ \bar{u} + .14\ o' + .17\ a \\ V_s &= .30\ i + .13\ o + .10\ a' + .04\ u + .31\ e + .02\ y + .10\ w \\ C_n &= .25\ r + .24\ l + .08\ ng + .26\ n + .17\ m \\ C_s &= .14\ d + .31\ t + .08\ b + .10\ p + .09\ g + .18\ k + .03\ j \\ &\qquad + .03\ ch + .04\ p \\ C_f &= .10\ z + .32\ s + .10\ v + .15\ f + .02\ zh + .11\ sh \\ &\qquad + .06\ th' + .07\ th + .07\ st. \end{aligned}\right\} \quad (10)$$

The sound index is related to these values by

$$i = .162\ V_l + .196\ V_s + .255\ C_n + .252\ C_s + .135\ C_f. \quad (11)$$

For obtaining the most representative single value for the syllable index I, the equation given below is used.

$$I = .5\ i^2 + .5\ i^3. \quad (12)$$

This equation is based upon the frequency of occurrence of the syllable forms in English speech. As pointed out before, if the compound consonants be considered as simple sounds, then there are less than 10 per cent of syllable forms other than the two- and three-sound type. The frequency of occurrence of these two types is approximately equal.

Similar formulæ to the above may be used to relate articulation results in English to articulation results in a different language. To do this it is necessary to select the fundamental sounds of the different languages that correspond to the 36 fundamental sounds of English, where the correspondence is based on similar phonetic characteristics and similar positions of the vocal organs in producing the sounds. When this is done, the coefficients in Eqs. 10, 11 and 12, must be modified to correspond with the frequencies of occurrence of the sounds and syllables in the language.

Observed values of individual sound articulation are thus reduced

to a single index, or for a more comprehensive picture, to five indices corresponding to the five groups of speech sounds. In order to compare the indices obtained by a given crew with those of a reference crew, it is necessary to correct the data in accordance with Eq. 2 for the effects of practice. To do this, as previously discussed, articulation tests are made upon one or more of the reference circuits by the crew in question. If I' is the syllable index so obtained, the practice factor for the crew is given by the relation

$$x = \frac{\log (1 - I')}{\log (1 - I)} . \tag{13}$$

The practice factors for the other indices may be obtained also, by substituting the appropriate indices for the syllable index in Eq. 13. In Table XV the reference values for the various indices are given for the reference circuits that were previously described.

TABLE XV

REFERENCE VALUES

Circuit	V_l	V_s	C_n	C_s	C_f	i	I
Master Reference System	98.5	98.9	99.6	99.2	98.8	99.3	98.0
Auxiliary Circuit of Master Ref. Sys.	95.0	95.0	96.5	88.5	66.5	90.0	77.0
Carbon Transmitter Circuit	97.0	97.0	96.5	93.5	82.0	94.0	85.5
Master Ref. Sys. plus 3,750- L.P.F.	99.0	99.5	99.5	99.0	86.5	97.6	94.0
" " " " 750- H.P.F.	96.0	92.5	99.0	99.5	98.5	97.1	93.0
" " " " 1,500- L.P.F.	93.5	86.5	90.5	76.0	52.5	80.2	58.0
" " " " 1,500- H.P.F.	85.0	82.5	96.0	97.5	97.0	92.0	81.0

If the values for the sound index be compared with the sound articulation values based on uniform weighting, that were given under the section on practice effects, it will be seen that for these circuits there is very little difference, between the two sets of values. In other words, the average sound articulation is very nearly equal to the average that is obtained when the individual sound articulations are weighted according to the frequencies of occurrence of the sounds in English.

Similar comparisons have been made for a large number of other transmission systems. They showed similar small differences between the weighted and unweighted averages. For this reason we consider it unnecessary to use the weighted average when great accuracy is not required, for example, in a great deal of our routine work where comparisons are being made between circuits which have similar characteristics. This means that when testing an unknown circuit,

having an electrical characteristic similar to one of the reference circuits, the syllable index I can be calculated from the observed syllable articulation S (as obtained with the new standard lists) by means of the equation,

$$I = .5\ S^{2/3} + .5\ S. \qquad (14)$$

This value must now be reduced to the reference condition of practice by the methods which have already been described. In such cases it is thus possible to obtain the syllable index from the observed syllable articulation values, and it is unnecessary to analyze the data for the individual sound articulation values.

The weighted average, however, is the more logical way of obtaining a single index and should be used when it is suspected that it might give results which are essentially different from the unweighted average.

It is possible to carry the probability relations a step further and apply them to cases of English words and sentences. In order to do this it is necessary to make assumptions as to how the thought or meaning of the words affects the interpretation of the sounds. These assumptions are not only somewhat uncertain, but owing to psychological factors in testing are difficult to verify experimentally. In general, the meaning associated with words makes them easier to interpret than meaningless words. For single-syllable words, these effects are small. Two-syllable words are easier to interpret than single-syllable words. The interpretation of words containing from three to five syllables, and short sentences, depends almost entirely upon interpreting those parts which are not indicated by the thought or meaning.

Other Testing Methods

For most articulation studies it has been found desirable to use the standard testing technique which has been described, but it is frequently necessary, in special cases, to use other techniques. In such cases it is desirable, if possible to interpret the results in terms of the standard technique. In the course of research work, several different articulation testing methods have been used which give information on the type of correlation between them that may be expected.

The probability relations have been made use of in constructing two other types of lists which are called vowel-consonant and vowel word-consonant word lists. These lists are designed to give the same values of sound articulation as given by the standard lists. The former lists are shown in Table XVI. The various vowels are combined with the same consonant, and the various consonants with the

TABLE XVI

Vowel List

Sound to be Graded	Testing Syllables in the List	
a	at	ta
ā	āt	tā
a'	a't	ta'
e	et	te
ē	ēt	tē
i	it	ti
o	ot	to
ō	ōt	tō
o'	o't	to'
u	ut	tu
ū	ūt	tū

Consonant List

Sound to be Graded	Testing Syllables in the List					
b	bū	ūb	ba	ab	bē	ēb
d	dū	ūd	da	ad	dē	ēd
f	fū	ūf	fa	af	fē	ēf
g	gū	ūg	ga	ag	gē	ēg
k	kū	ūk	ka	ak	kē	ēk
l	lū	ūl	la	al	lē	ēl
m	mū	ūm	ma	am	mē	ēm
n	nū	ūn	na	an	nē	ēn
r	rū	ūr	ra	ar	rē	ēr
p	pū	ūp	pa	ap	pē	ēp
s	sū	ūs	sa	as	sē	ēs
sh	shū	ūsh	sha	ash	shē	ēsh
th'	th'ū	ūth'	th'a	ath'	th'ē	ēth'
th	thū	ūth	tha	ath	thē	ēth
t	tū	ūt	ta	at	tē	ēt
v	vū	ūv	va	av	vē	ēv
ch	chū	ūch	cha	ach	chē	ēch
z	zū	ūz	za	az	zē	ēz
j	jū	ūj	ja	aj	jē	ēj
h	hū		ha		hē	
w	wū		wa		wē	
y	yū		ya		yē	
zh		ūzh		azh		ēzh
ng		ūng		ang		ēng
st		ūst		ast		ēst

same vowel. The technique of using the list is the same as that previously described, except that the vowel articulation and consonant articulation are measured separately. Only the vowel errors are counted when using the vowel list and only the consonant errors when using the consonant list. These lists have the advantage that they can be used over and over by merely changing the sequence of the syllables.

Table XVII shows two lists similar to the above except that they are made up entirely of common English words. They are designated as vowel word and consonant word lists. This list is used in the same way as the vowel and consonant lists. In using either of these lists

the testing crews should be familiar with the syllables or words in the lists.

TABLE XVII

VOWEL WORD LIST (ENGLISH WORDS)

Sound to be Graded	English Words in the List	
a'	bat	back
ā	bait	bake
e	bet	beck
ē	beat	beak
i	bit	bit
ī	bite	bike
o	but	buck
o'	bought	balk
ō	boat	boat
u	book	book
ū	boot	boot

CONSONANT WORD LIST (ENGLISH WORDS)

Sound to be Graded	English Words in the List	
b	by	by
ch	which	which
d	die	die
f	fie	whiff
g	guy	wig
h	high	high
j		
k	wick	wick
l	lie	will
m	my	whim
n	nigh	win
ng	wing	wing
p	pie	whip
r	wry	wry
s	sigh	sigh
sh	shy	wish
th'	thy	with
th	thigh	thigh
t	tie	wit
v	vie	vie
w	why	why
y		
z	whiz	whiz
st	sty	whist

Note: The h following w is not pronounced in such words as whim, whip, etc.

It usually requires a training period of a month or more for a testing crew to thoroughly master the technique of using the standard lists, that is, to reach a stage where the phonetic symbols are spoken and recorded almost mechanically. The vowel consonant lists require less time, since it is only necessary for the observers to fix their attention on one sound in the syllable. With the word lists this training period is reduced to a minimum. Phonetic symbols are avoided, and attention is given to only one sound in the words. As may be seen from Fig. 2, after a few tests they practically reach a degree of

uniform proficiency. In using the lists, the words are recorded with the English spelling. Only errors in the vowel and consonant sounds of the left-hand column of the above table, are counted. Since only one sound in each syllable is utilized, the above lists require a somewhat greater testing time for a given precision than do the standard lists where all three sounds of the syllables are used.

Table XVIII below, shows data that were obtained with the three types of lists, namely, the standard lists, the vowel consonant lists, and the vowel word consonant word lists. The vowel consonant

TABLE XVIII

ARTICULATION RESULTS WITH VARIOUS LISTS

Circuit	Cons. Vow. List		Cons. Word Vow. Word List		Standard List			Corrected
	C	V	C_w	V_w	S	VC^2	$V_wC_w^2$	$V_wC_w^2$
Air Transmission	99.5	99.0	99.0	99.5	98.0 ± .4	98.0	98.0	97.5
Master Ref. System + 5500 L.P. Fil.	99.0	99.0	99.0	99.5	96.5 ± 1.0	97.0	97.5	97.0
M.R.S. + 3750 L.P. Fil.	95.5	98.5	96.5	99.0	90.5 ± 2.0	90.0	92.0	90.0
M.R.S. + 1950 L.P. Fil.	82.5	98.5	85.5	98.5	67.0 ± 2.0	67.5	72.0	68.0
M.R.S. + 1000 L.P. Fil.	62.0	71.5	67.0	82.0	29.0 ± 3.0	27.5	37.0	33.0
M.R.S. + 1500 H.P. Fil.	96.0	81.0	97.0	86.0	75.0 ± 2.0	75.0	81.0	77.0
Carbon Transmitter Circuit	91.0	96.0	91.5	99.0	81.0 ± 1.5	80.0	83.0	80.0

lists lead to syllable articulations that are essentially the same as obtained with the standard lists. It is evident from the data that the word lists lead to slightly higher values of syllable articulation. The explanation is to be found in the make up of the word list. It was not possible to arrange the lists so that all sounds are equally probable, and still use English words. For instance, an observer would not record "bik" for "beck," nor "wiv" for "with," etc. For this reason, the observers do not make errors which occur frequently in the other two types of lists, where any sound is possible. Hence the observed articulations are somewhat higher with the word lists.

It was found, however, that the word lists may be correlated with the standard lists by means of the relation given by Equation 1. The value of x for this case is 0.9 so that, the word technique may be corrected to the standard technique, by means of the equation.

$$S = 1 - (1 - V_wC_w^2)^{\cdot 9}, \tag{15}$$

where S = syl. art. of standard lists expressed as a ratio,

V_w = vowel art. of vowel word lists expressed as a ratio,

C_w = cons. art. of cons. word list expressed as a ratio.

The corrected values are also shown in the above table.

It is frequently necessary to test very poor systems where the standard lists giving an articulation of a few per cent, are not satisfactory. The vowel consonant lists are somewhat more satisfactory under these circumstances. Lists of sentences have also been found to be very useful for such purposes. The sentences were of the interrogative or imperative form containing a simple idea. They were designed to test the observer's acuteness of perception rather than his intelligence. Tests were made with these sentences and the standard lists on various circuits, involving carbon transmitter circuits and various filter systems. The data are shown in Fig. 11. The

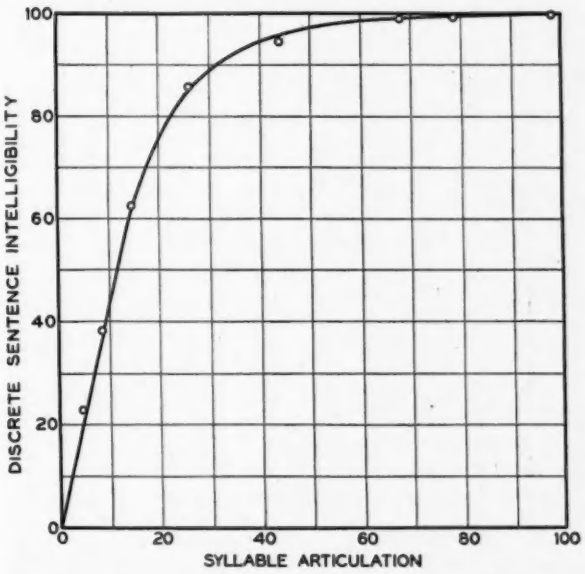


Fig. 11—Discrete sentence intelligibility vs. articulation

sentences were considered to be understood if the observer either recorded the sentence correctly or recorded an intelligent answer. As stated earlier, the percentage correctly observed is called the discrete sentence intelligibility.

It will be seen that for changes in distortion, the changes in the discrete sentence intelligibility will be small for systems having syllable articulations greater than 30 per cent, but very large for systems having syllable articulations below 20 per cent. It is for systems in this latter class that these test sentences are useful. A case in point is the measurement of the degree of secrecy obtained

in sound proofing telephone booths, or in dealing with cross-talk. The sentences have also been found to be useful in making quick qualitative tests of the goodness of an audiphone set for a particular case of deafness.

Because of their general usefulness for these purposes, the complete lists of sentences are given in the appendix. Due to memory effects a set of sentences can be used with the same personnel only a very few times. The psychological factors are also more prominent with sentences than with simple syllable.

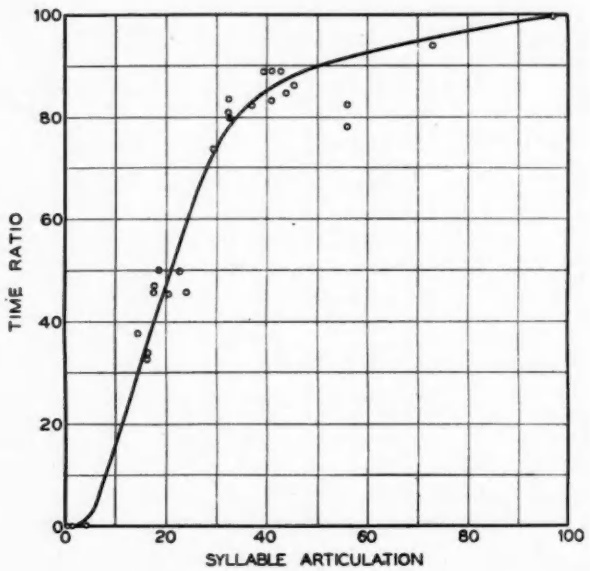


Fig. 12—Articulation vs. time ratio

Sentence lists of the above type have also been used to obtain a notion of how the time taken to transmit an idea correctly over a system depends upon the articulation. To do this, the observer was instructed to reply orally to the question. If the reply indicated that the observer failed to understand, the speaker repeated the question. Both speaker and observer tried to carry out the test in a normal conversational manner. The observer could ask the speaker to repeat, reword or spell out difficult parts of the sentence.

The tests were made on a variety of systems of known syllable articulation. The results that were obtained are shown in Fig. 12. The ordinates of the curve give the ratio of the time required to

transmit correctly one of these test sentences over an ideal system to the time required over the system under test. With the crew used in making these tests, and with an ideal transmission system, it required an average time of 5.2 seconds after the speaker started to pronounce the sentence before the observers grasped the idea. It will be seen from the curve that for systems having approximately 20 per cent articulation, the time required is twice as great. Fig. 11 shows that one out of every four of the sentences is mistaken for this value of articulation. If it is assumed that an observer asks that only sentences which he fails to understand be repeated, it can be shown that this time ratio is equal to the discrete sentence intelligibility.[11]

It is evident from Figs. 11 and 12, that the observed time ratio is appreciably less than the discrete sentence intelligibility. This difference may be taken to indicate that an observer not only asks that sentences which he fails to understand be repeated, but also that sentences about which he is uncertain be repeated. In other words, the time element reflects both factors, the understandability and the uncertainty.

As has been previously mentioned, tests have been made with various types of English word lists. Because of the manner in which the words were selected, and also due to uncertain psychological factors entering into the tests when such words are used repeatedly, it is difficult to compare the results so obtained with syllable articulation results.

However, it was found that if a definite rule were followed in selecting words from a newspaper, consistent results could be obtained with lists containing 500 or more words per list. The method of selection was to take the first word from every third line of a newspaper column. In this selection all proper names and the following six most frequent words of English were excluded, the, of, and, to a, in. When a word was hyphenated from the previous line, the whole word was used. Each of eight callers called a list of 66 words to four observers in the manner of an ordinary standard articulation test. Tests were made with the carbon transmitter circuit and the six circuits indicated in Fig. 3. The data were analyzed to give the discrete word intelligibilities for the one, two, three, four, and five-syllable words occurring in the lists, as well as for the lists as a whole. The lists on the average contained 46.3 per cent one-syllable, 29 per cent two-syllable, 16.8 per cent three-syllable, 6.4 per cent four-

[11] "A Theoretical Study of Articulation and Intelligibility of a Telephone Circuit," John Collard, *Electrical Communication*, 7, page 168, January, 1929.

syllable, and 1.5 per cent five-syllable words, and an average number of two syllables per word. The discrete word intelligibility vs. syllable articulation as obtained with the standard lists is shown in Fig. 13. The dashed curves indicate the relations for the various types of words, and the solid curve for the word lists as a whole. The data for two syllable words practically coincided with the solid curve. Owing to the small amount of data, the curves for the four- and five-syllable words are less reliable than those for the other types. Curves of the above type, both for words and sentences, depend very

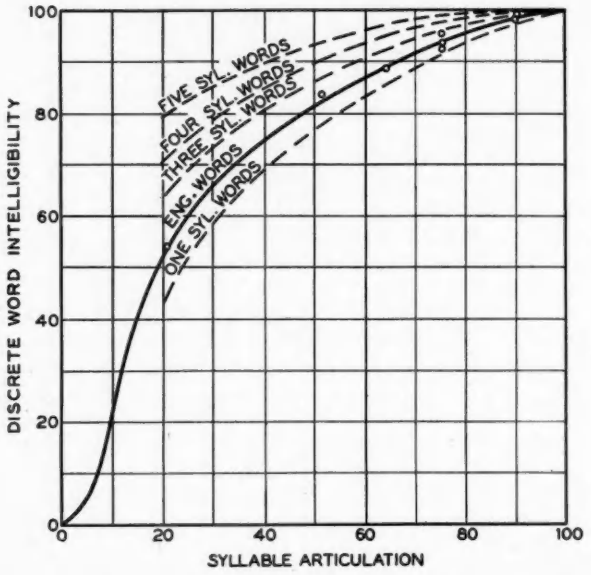


Fig. 13—Discrete word intelligibility vs. syllable articulation

much upon the way the speech material is selected. If, for example, only "different" words had been included in the word lists, appreciably smaller values of discrete word intelligibility would have been obtained.

Tests have also been made with lists made up of the following numbers, 1, 2, 3, 4, 5, 6, 8. These numbers were combined at random into groups of three and called in the manner of an ordinary articulation syllable. The distinguishing characteristic of each of the above numbers is a vowel sound, so that, they are interpreted primarily from recognizing the vowel. Such lists, therefore, do not give a very good picture of the speech capabilities of a system which distorts speech. They are, however, very useful in measuring the deafness

of an observer, for the reason that the number articulation decreases very rapidly as the sounds approach the threshold of hearing. As may be seen from Fig. 14, the number articulation passes from practically 100 per cent to 0 per cent in the short range of 10 or 15 db. It is evident that such lists give a critical measure of the point at which

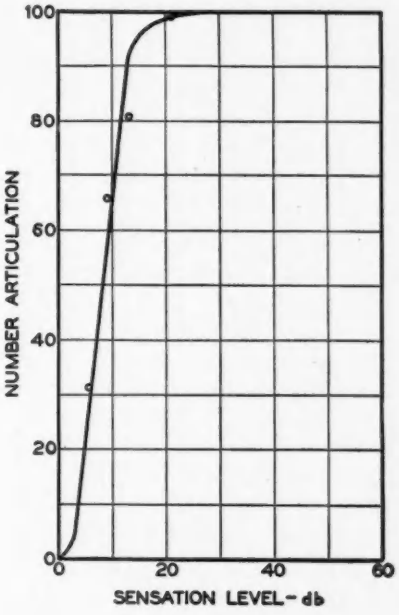


Fig. 14—Number articulation vs. sensation level

an observer fails to hear the sounds. Lists of this type have been used extensively in testing the hearing of school children.

Summary

The standard testing technique is primarily a means of determining the articulation or recognizability of the individual speech sounds when they are spoken in a way that is representative of conversational speech, and in a way which facilitates the carrying out of articulation tests. The articulations of the individual sounds may be converted into an index which indicates the speech capabilities of a system. Other types of lists which yield either the recognizability of speech sounds, or the intelligibility of discrete English words and sentences containing thought, have been described and experimentally correlated with the syllable technique.

It should be emphasized that there may not be a one to one corre-

spondence between all of these measured quantities for all types of speech distortion. The data entering into the sound, vowel, consonant, and syllable articulation curves were very extensive with respect to types of distortion and testing personnel. The theoretical equations relating them seem to rest upon assumptions with few uncertainties. For these reasons, it is felt that these relations can be used with considerable confidence, especially for values of syllable articulation greater than 30 per cent. The curves dealing with English words and sentences are based upon less diversified data and should be regarded as indicative only of the correlation and the type of relation.

During the past few years articulation testing methods have been used more and more, both in this country and abroad. It is felt that in order to compare the results obtained by various crews in various tongues, *it is desirable to use techniques that operate on the same basic principles and to calibrate various crews on similar reference circuits.*

INTELLIGIBILITY LIST 1

1. Name a prominent millionaire of the country.
2. How large is the sun compared with the earth?
3. Why are flagpoles surmounted by lightning rods?
4. Give the abbreviation for January and February.
5. Name the tree on which bananas grow.
6. How often does the century plant bloom?
7. What description can you give of the bottom of the ocean?
8. Explain the difference between a hill and a mountain.
9. What is the chief purpose of industrial strikes?
10. Describe the shoes of the native Hollander.
11. Name some uses to which electricity is put.
12. What would cause the air to escape from a bicycle tire?
13. Where is more grain raised, in the East or the West?
14. Tell what is meant by an Indian Reservation.
15. For what invention is Thomas Edison noted?
16. Name a state which has no seacoast.
17. Write the Roman numeral ten.
18. Explain the difference between export and import.
19. Explain why a corked bottle floats.
20. What substance is a good conductor of electricity?
21. Explain why Indians were afraid of firearms.
22. Explain the purpose of fire drills.
23. At what time do ocean waves become dangerous?
24. What medicine would you take to remedy indigestion?
25. What knowledge is covered by the study of astronomy?
26. Name a good restaurant in this vicinity.
27. What is the importance of large windows in stores?
28. Explain why a giraffe eats the foliage of trees.
29. How are the pages of a magazine held together?
30. Explain why the name string-bean is appropriate.
31. Name a nearby city in which there is a shipyard.
32. Name a fruit which grows in bunches.
33. Which of our Presidents went to South Africa?
34. Why are wire springs used in beds?
35. Why are books bound in stiff covers?
36. Why did the home people conserve food during the war?
37. Name an insect that has a hard shell.
38. What symbol on the United States money stands for liberty?
39. What weapons did the Indians use in warfare?
40. In what kind of weather does milk sour?
41. What streets in this city have Dutch names?
42. How does turning a ship's wheel steer the ship?
43. What nation aided us in the Revolutionary War?
44. What are some personal characteristics of the people of Japan?
45. What candy is black and good for colds?
46. Name a famous Indian Tribe.
47. Why is this building lighted by reflected light?
48. Why are most lighthouses situated on rocks?
49. Give some ingredients used in soap.
50. Why is a house built of stone superior to others?

INTELLIGIBILITY LIST 2

1. What recent epidemic overspread the world?
2. With what instrument are corks drawn from bottles?
3. What materials are used for the rims of eyeglasses?
4. In what country is sugar cane raised extensively?
5. What material does a swallow employ for its nest?
6. Name three mineral products.
7. Why is it dangerous to stand under a tree during a thunder storm?
8. Give four colors in the rainbow.
9. What do you consider a living wage?
10. Discuss the meaning of the word "temporary."
11. Explain what is meant by the term "trucking."
12. Mention some articles on which a war tax is levied.
13. How long a vacation did you get last year?
14. For what invention is Robert Fulton famous?
15. Which is the taller tree, the oak or the apple?
16. What woman was burned at the stake for witchcraft?
17. Name three precious stones.
18. To what island are immigrants taken when they come to America?
19. Do you prefer to draw with a ruler or free hand?
20. To what countries are missionaries sent today?
21. What is the significance of a dog wagging his tail?
22. What do you think is the work of an inspector of post offices?
23. What causes a person's nose to become red?
24. Briefly describe the American flag.
25. Of what material are men's collars made?
26. From what bird are ostrich feathers obtained?
27. Why is Mexico not a well-governed country?
28. Name a group of mountains in New York State.
29. Why were ancient castles surrounded by a ditch?
30. Tell in what way a zoo is beneficial to children.
31. Why aren't objects hurled from the earth by its rotation?
32. Why do we put awnings on our windows in summer?
33. What will a rat biscuit do to a rat?
34. Why is it dangerous to throw away a lighted match?
35. On what kind of plant do dates grow?
36. How often have you wished you could play the violin?
37. What work is done by negroes in the South?
38. Name the plant from which sugar is manufactured.
39. What conclusion do you use for a business letter?
40. What use would a carpenter make of a ladder?
41. Why do we have bars in front of jewelers' windows?
42. In what way is the horse a valuable animal?
43. Why are lockers provided for our hats and coats?
44. Where did you spend your vacation last summer?
45. What countries are known as the United Kingdom?
46. Why is the Christmas tree called an evergreen?
47. What country is represented by the stars and stripes?
48. Which is the most popular newspaper in the metropolis?
49. What do the Gold Dust Twins represent?
50. Name a common wild flower.

INTELLIGIBILITY LIST 3

1. What is meant by "A stitch in time saves nine"?
2. What is the first letter of your last name?
3. Why is there a spring in a window shade roller?
4. What is meant by the expression "during rush hours"?
5. How many judges make up the supreme court?
6. Why is it necessary to build foundations for houses?
7. Name the tool with which a burglar opens a window.
8. Of what benefit was the Red Cross to soldiers during the war?
9. What man is called the "Father of his country"?
10. What instrument do we use to drive nails into wood?
11. How many miles do you travel daily?
12. Three inches is what part of one foot?
13. Name a large steamer that sails from this port.
14. What substance is employed for the manufacture of lead pencils?
15. Name some furniture made from pine.
16. What is used at home parties for dance music?
17. Why does oil float on the surface of water?
18. What flowers float on the top of the water?
19. What harm is done by waves at sea?
20. What do we mean when we say that the storm has abated?
21. How does a man differ from other animals?
22. Mention a winter sport of Canada.
23. Why is it desirable to paint an iron fence?
24. Where does a sailor spend most of his time?
25. What is the principal beauty of Niagara Falls?
26. Of what advantage is a hooked beak to a parrot?
27. Of what assistance to the duck is its webbed feet?
28. How many of your grandparents are living at present?
29. Why do people wear low shoes in summer?
30. Describe the appearance of a falling star.
31. Make a list of articles made of wool.
32. Why do we use more coal in winter than in summer?
33. Why is it dangerous to fall from a great height?
34. Name an organization for the benefit of the soldiers.
35. What is the color of the suit worn by policemen?
36. Why are electric lights covered by frosted shades?
37. When are roof gardens more comfortable than theatres?
38. What can be done to avoid teeth decaying?
39. In what way will prohibition benefit drunkards?
40. Name an animal that lives on vegetation.
41. For what purpose are life-buoys placed in the water?
42. How many hours sleep did you have last night?
43. Name a large hotel in New York City.
44. What time of the year are parsnips planted?
45. What is the purpose of putting knobs on doors?
46. What city gets notoriety for baked beans?
47. Name a state belonging to New England.
48. What kind of shoe polish do you use?
49. Mention some important part of a lamp.
50. Why do men carry matches in their pockets?

INTELLIGIBILITY LIST 4

1. Name three modes of travel today.
2. State in what occupation the compass is indispensable.
3. What would you call a person who shrinks from danger?
4. What tropical fish has a sword for protection?
5. Name some different kinds of cheese.
6. Name the chief food of a horse.
7. What cathedral was destroyed during the last war?
8. Explain why geese hiss when angry.
9. Where is tobacco cultivated in the United States?
10. Name the national anthem of France.
11. Give the reason for winding a watch.
12. Name a weed that sticks to your clothing.
13. In what part of this country did the Quakers settle?
14. Why is it necessary to fill out time cards?
15. Of what material are the sails of a ship made?
16. Why is the climate warmer in California than in New York?
17. Explain why you enjoy resting by a stream.
18. Which flies by night, the butterfly or the moth?
19. Explain why a zeppelin floats in air.
20. Name two newspapers published daily.
21. From what deep sea animal is tortoise-shell obtained?
22. Why do engineers draw plans of machinery?
23. How can a ship's crew escape from a wreck?
24. By what means does a snake protect itself?
25. Where do people in cities spend their vacations?
26. From what plant is cotton-seed oil manufactured?
27. Explain the necessity for docks in a harbor.
28. What is the reason for painting frame houses?
29. What metal is used in making watch springs?
30. From what ingredients is sand paper made?
31. What do the American people consider the staff of life?
32. In what region is ranching the principal industry?
33. Explain why a humming bird has a long bill.
34. In what month do we celebrate Thanksgiving Day?
35. What tools would you use to make up a tool box?
36. How did the Dutch buy land from the Indians?
37. Near what city is the Statue of Liberty situated?
38. Why do people keep gold fish in globes?
39. State the necessity for fly screens.
40. Contrast the size of a grape fruit and a grape.
41. Compare the prices of 1916 and those of today.
42. What machine is used in drawing water from a well?
43. What dress should be worn to a formal dance?
44. What president read the "Proclamation of Emancipation"?
45. What company makes fifty-seven varieties?
46. What is the difference between a wild animal and a tame one?
47. Tell me where an eagle builds its nest.
48. Explain why we wear light colors in summer.
49. Describe what the Indians used as a head-dress.
50. Where are some of the highest mountains in the United States?

INTELLIGIBILITY LIST 5

1. Why are book shelves placed in a library?
2. Into what body of water does the Hudson empty?
3. Name the good roads built by a state.
4. Which is more common, a three or four leaf clover?
5. What is the chief business of the Chinese in this city?
6. What tool is used by a brick mason?
7. What is the advantage of belonging to a public library?
8. Give a reason for building fountains in parks.
9. What is the answer called when we subtract numbers?
10. What large vessel was sunk by a German submarine?
11. Does poison ivy grow on a tree or a vine?
12. Which travels faster, an aeroplane or a locomotive?
13. What dangers do vessels meet on the ocean?
14. Why are clothes closets built in houses?
15. What is the most interesting novel you have read?
16. Why do you call a plumber when a pipe bursts?
17. What brand of canned soup do you take camping?
18. Name your favorite griddle cakes for breakfast.
19. To what city do the Manhattan and Bronx boroughs belong?
20. What large animal does the heavy work in India?
21. Why are lightning rods used on barns?
22. In what part of the sky does the sun appear to set?
23. What does "P.S." mean on a school building?
24. What kind of boat first crossed the ocean?
25. What happens when the sun goes behind a cloud?
26. How is the ground on a farm prepared for planting?
27. Name an insect which carries disease.
28. What tree is a source of joy December 25th?
29. Draw a picture of a leaf of the cornstalk.
30. Why is the mosquito considered an undesirable insect?
31. Name the present king of England.
32. Which is more strenuous, walking on the level or uphill?
33. Tell the country to which Caruso belongs.
34. Which plant grows higher, a hollyhock or a pansy?
35. Why do street cars stop at every corner?
36. Mention a country whose civilization is of low grade.
37. Name the President of the United States.
38. What happened to the Hamburg-American Line?
39. Name the three meals we have a day.
40. Explain how we can tell when the wind is blowing.
41. Why is marble used for ornamental purposes?
42. Name our famous geyser park.
43. Distinguish between the use of the singular and plural.
44. Name some important parts of the telephone.
45. What are two products secured from the olive tree?
46. Explain the reason for growing ferns in our homes.
47. Are the soles of your shoes sewed or nailed?
48. How can roads help to prevent famine?
49. What machine is used to cut lawns?
50. Why did the southerners want to keep slaves?

INTELLIGIBILITY LIST 6

1. Why do cities provide popular concerts?
2. Why don't you stop coffee and drink postum?
3. At what place does the President of the United States live?
4. Name some article which may be sent by Parcel Post.
5. Why are summer suits called "Palm Beach"?
6. Briefly describe the plumage of the peacock.
7. What is a safe rate of speed for an automobile to travel?
8. What domestic fowls would you include under poultry?
9. Why is it dangerous for boats to sail in a fog?
10. Why should American homes possess an American flag?
11. In what country do rubber trees grow?
12. Why do modern road-builders cut through hills?
13. What is the advantage of using a derrick to lift rocks?
14. Name an eminent American poet.
15. What tool does a carpenter employ to saw a board?
16. Where does rice grow, in swampy or dry ground?
17. What did people formerly think about the form of the earth?
18. With what material would you stuff a bean bag?
19. Explain the use of waiting rooms in depots.
20. Why does the bee visit the fragrant flowers?
21. Through what profession do people get legal help?
22. What machine has replaced horses on farms?
23. Describe the appearance of the sky before a thunderstorm.
24. What cereal did you have for breakfast this morning?
25. Tell me why children love to gather flowers.
26. Who was the French president during the war?
27. How do we defend our country against foreign invasion?
28. Under what conditions is it possible to hear an echo?
29. What is the penalty paid by a spy when caught by the enemy?
30. Why is it necessary for men to study medicine?
31. What effect does a hot climate have on Eskimos?
32. Which do you prefer, a detective story or a romance?
33. Why does a woman wear a sweeping cap?
34. Name a useful animal for which Arabia is noted.
35. Explain the convenience of brown paper envelopes.
36. What can you write with besides a pen and pencil?
37. How can we tell when an aeroplane is near?
38. Why is a stool provided for motormen in street cars?
39. Describe the difference between rain and hail.
40. How many ounces equal two pounds?
41. Tell me the color of the pencil you are using.
42. Why are typewriters preferred for business letters?
43. What countries border the United States?
44. Mention some food that is salable at a public market.
45. Name the drug that is extracted from the poppy.
46. Name the homestead of George Washington.
47. Name the colors you see in a rainbow.
48. Name four fishes used for food.
49. Explain what Y. M. C. A. designates to the public.
50. Why do you prefer home-made bread to baker's bread?

INTELLIGIBILITY LIST 7

1. What are the largest lakes in North America?
2. At what time do people sleep, during the day or during the night?
3. Name the tree from which corks are obtained.
4. Name the country to which Alaska belongs.
5. Which game is more scientific, golf or croquet?
6. From what island does Manila rope come?
7. Who do children think brings presents at Christmas?
8. Name a cathedral that is noted for its architecture.
9. What animals in India draw carts?
10. Explain the difference between a drone and a working bee.
11. When are the days longer, in the summer or the winter?
12. Tell what is meant by poultry farming.
13. Where do you expect to spend your summer vacation?
14. Of what does the food of a night hawk consist?
15. What was the reason for camouflaging steamships during the war?
16. Name the country in which the Alps are situated.
17. Which resort do you prefer, mountain or seashore?
18. Explain the necessity for a mast on a yacht.
19. In what state was Woodrow Wilson born?
20. Why does the word cunning apply to a fox?
21. What is the busiest time of the year for farmers?
22. Explain the advantage of a city directory.
23. Tell me your two favorite sports.
24. Mention a brand of condensed milk.
25. What game bird makes delicious food?
26. Name the street where your home is situated.
27. What are some utensils necessary for cooking food?
28. Why is it necessary to wash glass windows?
29. Why is it necessary to have courts in large buildings?
30. Locate the financial street of New York.
31. How long must you wait for a messenger boy?
32. What is the place called where we get marble?
33. Which do you prefer, grand opera or comedy?
34. What benefit do plants derive from the sun?
35. What month of the year does February follow?
36. What is the importance of fire alarm boxes?
37. Tell me of what religious denomination you are a member?
38. Why should shoes have rubber heels attached to them?
39. Name the animal from which buckskin is made.
40. With what machine are heavy stones lifted?
41. What western state has been disturbed by earthquakes?
42. For what occupation are the banks of Newfoundland famous?
43. When do children make lanterns out of pumpkins?
44. On what avenue are most of the department stores situated?
45. What interesting novels have you recently read?
46. Make a list of products that the farmer raises.
47. What is the object of having public institutions for the helpless?
48. How would you ask for something to be passed at the table?
49. Tell in what way grass is useful.
50. What war was fought over the question of slavery?

INTELLIGIBILITY LIST 8

1. What part of the rhubarb plant is used for food?
2. Of what advantage are good roads to a country?
3. Which is greater, the population of New York or Boston?
4. In what four months is the letter "R" missing?
5. Why are there red and green lights on a ferry slip?
6. Name a river which separates New Jersey and Pennsylvania.
7. Explain why roots are necessary to a tree.
8. What story do you associate with the character Pocahontas?
9. At what time of the day is it warmer, morning or afternoon?
10. Of what materials are fountain pens made?
11. What is the means of transportation between New York and Staten Island?
12. Why do physicians use disinfectants on wounds?
13. Describe the locomotion of an inch worm.
14. Why is food kept on ice in summer?
15. On what part of the plant does an ear of corn grow?
16. What kind of boat is driven by oars?
17. Explain how butter is churned in large quantities.
18. Why is a daily paper called a newspaper?
19. By what means are pictures obtained for the movies?
20. By what event is the christening of a ship characterized?
21. Which are more precious, diamonds or rhinestones?
22. Tell what we do when we feel hungry.
23. Why do children enjoy a box of paints?
24. What kind of apparatus is used in a gymnasium?
25. By what means is hay lifted on top a stack?
26. Give a reason why it is so noisy in the city.
27. What substance is used for manufacturing candles?
28. What large city have you seen besides New York?
29. Name three of the thirteen colonies.
30. Which has more wheels, an automobile or a bicycle?
31. Why is it not safe to climb high?
32. Compare the postage on cards and letters.
33. Does water run uphill or downhill?
34. Is the Hudson River salt or fresh water?
35. What domestic pets do you keep at home?
36. What is meant by government ownership of railroads?
37. How do the horns of a sheep and goat differ?
38. Name a firm that manufactures chewing gum.
39. In what city is the Grand Central Station?
40. What day is considered the beginning of the week?
41. What kinds of shade trees are there in your community?
42. Compare the food value of radishes and meat.
43. Name the substance from which glass is made.
44. Which do you prefer, a bungalow or a city apartment?
45. How great is the water power of Niagara Falls?
46. What becomes of frogs and snakes during the winter?
47. By what means does a bird protect itself?
48. How did people travel before automobiles were invented?
49. How many fingers have you on each hand?
50. By what firm was your fountain pen made?

INTELLIGIBILITY LIST 9

1. From what animals are pork and leather obtained?
2. Name a large river in the United States.
3. Why are desk telephones the most convenient?
4. What is the difference between a dictionary and an encyclopædia?
5. What instrument is used for measuring temperature?
6. What implements are used for playing golf?
7. What instrument is used to unlock a door?
8. Give a brief description of an Indian.
9. Name three cities on the Atlantic Coast.
10. Name a plant which grows from a bulb.
11. Tell what harm can be done by icebergs.
12. What training is given to the Boy Scouts?
13. What private enterprises are now under government control?
14. How many pints are there in two quarts?
15. Spell the abbreviations for doctor and mister.
16. What is meant by a republican form of government?
17. Why are forts built near a harbor?
18. Describe the daily motion of the earth.
19. Name the cloth from which ships' sails are constructed.
20. Name the boroughs of New York City.
21. Why do motor trucks have broad wheels?
22. How can a fire be lighted without matches?
23. Name a baking powder universally known.
24. Why do farmers wear overalls when at work?
25. At what time does the sun rise in summer?
26. Which is more important to know, reading or dancing?
27. What punishment is inflicted upon a murderer in this state?
28. What beverages do you prefer to drink in summer?
29. Why do farmers employ sleds in winter hauling?
30. Give a description of a signet ring.
31. Why are bats used in a baseball game?
32. Locate a good southern harbor.
33. Why do we close the windows of a train going through a tunnel?
34. Give the color of the polar bear.
35. How are modern factories lighted in daytime?
36. What animal roamed in America at the time of the Indians?
37. Through what arithmetical process do we derive the quotient?
38. For what purpose are coat hangers used?
39. What is the color of the uniforms worn by soldiers?
40. For what purpose do people use highways?
41. Why does a fire-engine blow a whistle?
42. Of what material are Japanese lanterns composed?
43. Compare the color of carrots and beets.
44. What causes the majority of our railroad accidents?
45. Why is it bad to work after business hours?
46. Do you prefer to play football or baseball?
47. What bird can imitate the human voice?
48. What country has been devastated in the recent war?
49. What is used to stamp inscriptions on coins?
50. How is health affected by exercise and fresh air?

INTELLIGIBILITY LIST 10

1. Why do most people read the newspapers?
2. Name the fruit from which vinegar is obtained.
3. What historical scene would make an excellent picture?
4. Where do weeping-willow trees grow best?
5. Why are oars made flat at one end?
6. Why do farmers like to locate near cities?
7. Why is it annoying to handle a cactus plant?
8. In what way do cigarettes affect the health?
9. What do we call the building where automobiles are kept?
10. Why is it more difficult to talk in public than at home?
11. Compare the size of an apartment house and a bungalow.
12. Why should you avoid reading on a moving car?
13. Name the feature you prefer in a circus.
14. In what country did President Roosevelt go hunting?
15. Name some barnyard fowl you have seen.
16. How many seconds equal one minute?
17. What are the different ways of cooking eggs?
18. Why are most mantel-pieces handsomely carved?
19. Why do flowers droop in a heated room?
20. Tell in what state the Pilgrims landed.
21. Write down your last name and address.
22. Which metal is heavier, aluminum or iron?
23. Why are time cards essential for employees?
24. Why does the city clear the snow away?
25. Give some reason why air is important.
26. Which part of the United States has the most high land?
27. What power is used for propelling a motor-boat?
28. Suggest a scheme for earning money.
29. What is meant by the expression "a West Wind"?
30. Give the reason why a duck can swim.
31. Draw a sprig of pussy willow.
32. Describe how fire wood is secured.
33. Name four points of the compass.
34. Tell why the Puritans came to America.
35. Name the magnetic poles of the earth.
36. Which animal resembles man most closely?
37. Tell me how many brothers and sisters you have.
38. When does the tennis season begin and end?
39. Name the plant from which linen is made.
40. Explain the use of the periscope of a submarine.
41. What time of the year is harvest time?
42. How long does it take to cross the ocean?
43. Why does a physician count pulse beats?
44. What price do barbers charge for a haircut?
45. Write down the name of our national flower.
46. Is the silkworm a valuable or harmful insect?
47. What are the leading political parties of the country?
48. Mention a state in which women have the right to vote.
49. What organization nurses our wounded heroes?
50. For what is Wall Street world-wide noted?

INTELLIGIBILITY LIST 11

1. Explain the necessity for fire boats in the harbor.
2. Name an article made of tin.
3. What bird of Africa is noted for its plumage and speed?
4. Is Jupiter considered a large or small planet?
5. From what country are Turkish rugs imported?
6. Draw the shortest distance between two points.
7. Tell in what way telegrams are a benefit.
8. Name the large technical school in Boston.
9. To what part of the day does twilight refer?
10. Name a department store on Fifth Avenue.
11. Where did the Pilgrims get materials for their homes?
12. Near what country do we find icebergs?
13. Why are curtains and rugs put away for the summer?
14. Discuss the value of megaphones to speakers.
15. Give a reason why it is beneficial to have photographs.
16. Tell what we do to quench our thirst.
17. Give an example of evaporation that you have observed.
18. Where are the centers of the fishing industry in America?
19. What president's picture do we find on our dollar bill?
20. What vehicles are used for navigating the air?
21. What foods can be bought in a delicatessen store?
22. Why is it advantageous for office chairs to be adjustable?
23. What effect does early frost have on fruits?
24. What does the shape of a starfish suggest to you?
25. What race has been persecuted for its religious belief?
26. Explain the use of note paper.
27. Which is larger, a dreadnaught or a torpedo boat?
28. Which is longer, the minute hand or the hour hand of a watch?
29. What fowl is eaten by the Americans at Thanksgiving?
30. Name the continent to which Brazil belongs.
31. Explain why a dining table is an extension table.
32. What musical instrument is played by beating?
33. Estimate how many people patronize this cafeteria.
34. Name the country to which Canada belongs.
35. Draw a sign that represents a star.
36. Name a president of the United States whom you admire.
37. Name a state where an aviation field was located.
38. Explain why our forest land is diminishing.
39. At what time does a ship use its anchor?
40. What is a man who drives an automobile called?
41. For what reason are pages in books numbered?
42. What do you use to spread butter on bread?
43. On what street is the Western Electric situated?
44. What country produces olive oil?
45. Name an exercise in the water which is healthful.
46. Name an ink that cannot be erased.
47. What is the reason for irrigating arid land?
48. Do dates grow on a palm or pine tree?
49. Why does a grape arbor make an attractive summer-house?
50. Name the largest state in the Union.

INTELLIGIBILITY LIST 12

1. Why are water tanks built high?
2. Why are corks kept in ink bottles?
3. Name the factories for which Grand Rapids is famous.
4. What substances are attracted by a permanent magnet?
5. Tell me the purpose for which an ambulance is used.
6. What month of the year do you like best?
7. What are the red tanks on the tops of buildings?
8. Explain the duty of a train conductor.
9. Give the names of five vegetables.
10. What bird is engraved on the American dollar?
11. Tell me who darns and mends your clothes for you.
12. What are three diseases caused by germs?
13. Which contains more nourishment, pie or milk?
14. What stringed instrument is played with the fingers?
15. Give three historical events.
16. Compare the appearance of the crow and the robin.
17. What happens to the water in a brook when the sun shines on it?
18. Which is the prettier flower, the dandelion or carnation?
19. What was the ancient method of starting a fire?
20. Describe the fruit of the cherry tree.
21. Why are ambulances allowed to ring a gong?
22. Name four calculations that can be made on the comptometer.
23. What characteristic makes a porcupine disagreeable?
24. What African animal has black and white stripes?
25. Name some warm materials for clothing in the winter time.
26. Give five legal holidays.
27. Which receives the higher salary, the President or a Governor?
28. Where would we be most likely to find coral?
29. Where are box seats found in a theatre?
30. Name a museum where famous paintings are found.
31. Explain why you wear dancing pumps.
32. Why do men wear overcoats in the winter?
33. What great serpent infests the forests of Brazil?
34. Why is it dangerous for children to play with firecrackers?
35. Why do doors of public buildings swing outward?
36. What metal is used for the construction of tall buildings?
37. Do you prefer to travel by train or boat?
38. Compare the colors of raw and cooked meat.
39. Name the oldest bridge in New York.
40. How is the length of a piece of cloth measured?
41. What is the name of a plant covered with thorns?
42. Where are mosses and ferns found in abundance?
43. Give the initials of your full name.
44. How often do you go fishing each year?
45. What animal is known to sing by croaking?
46. Suggest some games to be played at a picnic.
47. Why do stores employ paper for wrapping packages?
48. Why are we prohibited from carrying dangerous weapons?
49. During what season does the farmer plow the fields?
50. In what part of America is cotton raised?

INTELLIGIBILITY LIST 13

1. What is meant by getting the sum of numbers?
2. What are two commercial uses for iron?
3. Name the building in which grain is stored.
4. What is built across a river to enable traffic to pass?
5. From what country is Swiss cheese imported?
6. Give a description of Haley's comet.
7. Of how many years does a century consist?
8. Explain why furnaces are installed in houses.
9. Of what advantage are long necks to swans?
10. From what skins do Indians make moccasins?
11. Tell me what seedless oranges are called.
12. Tell about the use of explosives in quarrying.
13. What use do people make of their teeth?
14. By what means are large portions of land measured?
15. Explain how Jersey milk is obtained in Ohio.
16. Name the baseball league of New York.
17. Explain what is meant by a civil war.
18. At what time of the year are the days and nights equal?
19. Explain why a squirrel buries nuts.
20. What is the proper way to ventilate a bedroom?
21. To what country or countries do you trace your ancestry?
22. Explain why ships carry wireless apparatus.
23. Name three fruits you like.
24. Name some spices used for seasoning.
25. What body of water separates Long Island from the shore?
26. What are the different ways of traveling about a city?
27. Give the name of your favorite college.
28. How is the fly able to walk on the ceiling?
29. Name the largest fish in the ocean.
30. In what part of the country are prairies to be seen?
31. Tell why dark glasses are used on beaches.
32. Where is the highest hill near your home?
33. What use do people make of rivers?
34. How may parcel-post packages be protected against loss?
35. Name a country where there are many police dogs.
36. What causes the wheel of a windmill to rotate?
37. Name a common shade tree.
38. How has this city been affected by New Jersey explosions?
39. By what power are trains brought into this city?
40. What term is applied to the abolishment of liquor?
41. By what means are clothes kept clean?
42. What useful work do dogs perform in Alaska?
43. What is the name given to the study of flowers?
44. What kind of work is done at a dry dock?
45. Why should children patronize public libraries?
46. What part of America did the Dutch settle?
47. Why are electric wires conveyed in lead cables?
48. What weather do dark clouds suggest?
49. What part of the day do we connect with dawn?
50. What is caused when two trains meet?

INTELLIGIBILITY LIST 14

1. When are Red Cross nurses needed in great numbers?
2. Mention a commercial use for isin-glass.
3. Name a steamship that belongs to the Cunard Line.
4. What articles are necessary for a fishing trip?
5. Where is the banjo the chief musical instrument?
6. Give the difference between a lemon and an orange.
7. What might happen to someone telephoning during an electric storm?
8. How many humps are on a camel's back?
9. What are two ways of heating buildings?
10. How does a mariner's compass indicate direction?
11. Why are electric signs used in advertising?
12. What is the better pen, Waterman or Conklin?
13. Name some important uses for rubber.
14. Mention an incident that happened to Robinson Crusoe.
15. Is slavery considered just or unjust today?
16. What street in this city is noted for displaying styles?
17. Why is Mothers' day observed in this country?
18. Why do children cry at bath time?
19. Give a word which rhymes with "fan."
20. Name one of England's famous queens.
21. Why did primitive people live in cliff dwellings?
22. Of what material are parquet floors made?
23. Name a popular play of the season.
24. What oceans touch the coast of North America?
25. What emotions can a dog express by barking?
26. Why is it valuable to big companies to send out catalogues?
27. What color are the stars on the American Flag?
28. Discuss the necessity for a ferry slip.
29. Tell why birds are frightened of cats.
30. Name and locate the capital of your state.
31. For what service are the students at Annapolis trained?
32. Name a great battle of the past war.
33. What stone is used for cutting glass?
34. Why is the moon generally invisible during the day?
35. Tell me your home telephone number.
36. Explain why farmers use scarecrows.
37. In what part of the country are the best apples grown?
38. What use do birds make of their bills?
39. Explain the necessity for a door knob.
40. In what town in Italy has there been a volcanic eruption?
41. What insect in New Jersey is widely known?
42. What kind of work does a draftsman perform?
43. What is the name of a bird used as a messenger?
44. Why was a canal cut through the Isthmus of Panama?
45. Which is the more expensive, cut or pressed glass?
46. What produce is raised on a truck farm?
47. What theatrical plays have you seen this year?
48. Tell how you would revive a person who has fainted.
49. What animals furnish fur for trade?
50. What is the penalty for objection to military service?

INTELLIGIBILITY LIST 15

1. Why is it dangerous to touch a live wire?
2. What advantage to the West is artificial irrigation?
3. Do you prefer soft gray or rich coloring?
4. On what occasion does a ferryboat whistle?
5. Why is it necessary to quarantine people with contagious diseases?
6. What things that your grocer sells come from afar?
7. What place have we for people who break laws?
8. Explain the use of hinges on a door.
9. What industry is carried on in the forests of this country?
10. On what occasion are flags at half mast?
11. Why do the majority of birds migrate in winter?
12. What work is accomplished in manual training schools?
13. What kind of shoes are worn to play tennis?
14. Compare the colors of the buttercup and clover.
15. What does pasteurizing do to the germs in milk?
16. Explain how we can tell when water is boiling.
17. Explain how the robin secures earthworms.
18. Why are electric wires put underground in cities?
19. What is a famous electric sign on Broadway?
20. From what material is a tennis racquet made?
21. Who was the first commander-in-chief of the American Army?
22. Write down the Arabic numerals from one to ten.
23. Why is hot water better for washing than cold?
24. What color is the fire-chief's automobile?
25. Of what benefit are parks in large cities?
26. Discriminate between the words optimistic and pessimistic.
27. Explain how to pull tacks from a carpet.
28. In what year was the Declaration of Independence signed?
29. Of what material are window screens made?
30. How many times a year do violets bloom?
31. What machine is used to write letters rapidly?
32. At what time do most business houses close?
33. Why are tacks used in laying a carpet?
34. What vines are useful for shading a home?
35. What bark do Indians use in making canoes?
36. Explain why elephant tusks are valuable.
37. Which do you prefer, chocolate or maple fudge?
38. What was Hoover's policy during the war?
39. Describe some animal found in the Rocky Mountains.
40. Name the most conspicuous object in the bay.
41. Which card game do you frequently play?
42. Name two kinds of nuts.
43. What is the color of the sky on a clear day?
44. Why do civilized people believe in prohibition?
45. What name is applied to a baby bear?
46. What is the easiest way to extinguish a fire?
47. What periodical is published by this company?
48. By what means can we move from one house to another?
49. Why do we walk more slowly uphill than down?
50. What country nearly supplies the world with coffee?

INTELLIGIBILITY LIST 16

1. What railroad connects New York and Chicago?
2. Name some benefits a Boy Scout receives.
3. At what place is the Capital of the United States of America?
4. What do physicians use to photograph parts of the body?
5. Name a place of interest at Washington, D. C.
6. Name the color of an orange skin.
7. Which is the more expensive, a Packard or a Ford?
8. Mention two beaches near the city.
9. For what purpose do children use pumpkins at Hallowe'en?
10. Name three different liquids.
11. In what part of the country is the Indian reservation?
12. Tell why tobacco is injurious to health.
13. Name the last war in which America participated.
14. Which is preferable, gas or electric light?
15. Why is summer school held at Columbia University?
16. Describe the color of a dogwood blossom.
17. Why are chimneys built on the outside of houses?
18. How does a reading lamp receive its name?
19. How many blades has a jack-knife?
20. What color is the water of the Mediterranean Sea?
21. Why do ships have their names painted on the stern?
22. Of what advantage is wireless to a ship at sea?
23. Why are forts unnecessary between the United States and Canada?
24. How do people protect their homes against burglary?
25. Explain how to magnetize your knife blade.
26. For what reason is a candle placed in a candle-stick?
27. Why do sea-going vessels carry life-boats?
28. What part of the world's sugar does Cuba supply?
29. Name five presidents of the United States.
30. Describe the color of a dragonfly's wings.
31. Name a country in which a king reigns.
32. What trade is principally carried on in Alaska?
33. Name a machine for lifting heavy weights.
34. State the duty of a vessel's pilot.
35. Tell briefly why mosquitoes are harmful.
36. Draw the dwelling place of an Indian.
37. Name the fruit from which grapejuice is made.
38. Name a state where grape vineyards are found.
39. By what means is the lawn-mower in the park propelled?
40. How do bees get pollen from flowers?
41. Which has the stronger odor, garlic or onions?
42. Name some of the articles manufactured from leather.
43. Name two of America's most wealthy men.
44. Why do we cook most articles of food?
45. What does a farmer plant after cultivating the ground?
46. Give me a word that rhymes with park.
47. What people are noted for their artistic ability?
48. Compare the weight of a quarter and a feather.
49. What registers the number of miles on an automobile?
50. Locate the main branch of the public library.

INTELLIGIBILITY LIST 17

1. What automobile tire is highly advertised?
2. Do you prefer peanuts raw or roasted?
3. What is the motive power of a sail boat?
4. Name the flower bulbs that are imported from Holland.
5. Name a liquid used for writing.
6. From what bird are plumes obtained for millinery?
7. How large does the full moon appear to you?
8. Which language is the most used in the world?
9. What material is used to cement bricks?
10. Name a distinguished statesman of today.
11. How many inches equal one yard?
12. What is the reason for so vast a population in New York?
13. What do most people work for, money or experience?
14. Which is more picturesque, a hilly or flat country?
15. Why do we use soap when we wash clothes?
16. In what month are apple blossoms in bloom?
17. At what time can owls see best?
18. What time of the year are most fruits ripe?
19. Of what material are the soles of arctics made?
20. Which do you like better, ice cream or soda?
21. Why is the dishwashing machine a great invention?
22. Name five cities in the United States.
23. What is a common means of travel for pleasure?
24. Of what advantage is a metronome to one studying music?
25. How far up the Hudson can river boats navigate?
26. How would you formally address an unmarried woman?
27. What day of the week is named after the sun?
28. Locate a large hotel in this city.
29. Name an explorer of the arctic regions.
30. Tell what coat hangers are made of.
31. What bait do fishermen use for trouting?
32. What happens to the Ohio River when the snow melts?
33. How did Abraham Lincoln get his education?
34. How much more is five than three?
35. Where do you order your stationery supplies in this building?
36. Name a beautiful monument you have seen.
37. What river separates the United States and Mexico?
38. Why does the law require dogs to be muzzled in public?
39. What kind of grass grows in a marsh?
40. Describe the difference between thunder and lightning.
41. What is the largest movie house in New York?
42. Explain why rubbers keep your feet dry.
43. By what method do fishermen catch fish for market?
44. Name a play written by Shakespeare.
45. From what substance is cream cheese made?
46. From what food product is molasses obtained?
47. Explain why a bugle is necessary in the army.
48. Why is New York often called the "Hub"?
49. How does the Leviathan compare with the Mauretania in size?
50. Explain the purpose for which a dictionary is used.

INTELLIGIBILITY LIST 18

1. In what state are oranges raised abundantly?
2. Name the man who manufactures the Ford automobile.
3. Describe the effect of heat on rice.
4. Where do the immigrants land in New York Harbor?
5. Who is the inventor of wireless telegraphy?
6. Tell what is meant by the word altitude.
7. Compare the temperature on mountains and in valleys.
8. Give the use of the transmitter of a telephone.
9. Explain how a bungalow differs from a house.
10. What liquid is mixed with paint before using?
11. What instrument does a man use for shaving?
12. Contrast the markings of a leopard and a tiger.
13. What material is used to stuff pillows?
14. Do you prefer to swim in fresh or salt water?
15. Does a sparrow sleep on the ground or in a tree?
16. What part of the plant grows underground?
17. Where do we find more negroes, in the North or South?
18. From what feathers are down pillows made?
19. Which has the better food value, meat or candy?
20. Discuss the necessity of a paper weight.
21. Of what use are elevators in high buildings?
22. Is bituminous coal hard or soft?
23. What does a cowboy use to lasso a buffalo?
24. What power runs a modern sewing machine?
25. On what island is the Western Electric situated?
26. Of what importance are refrigerators in hot weather?
27. Why is it dangerous to cross the Hudson in a canoe?
28. Explain why colleges encourage athletics.
29. Explain the need for the traffic police.
30. Name the discoverer of the Hudson River.
31. Give the chief food of the Chinese.
32. From what nations is the yellow race formed?
33. When is yawning and stretching considered rude?
34. Name some fish that live in fresh water.
35. For what reason is a tail attached to a kite?
36. Why do we swing our arms when we walk?
37. Why is the twelfth of February celebrated as a holiday?
38. In what ocean are the Philippine Islands situated?
39. Explain the daylight saving plan.
40. What country worships idols instead of God?
41. What city has the largest population in the United States?
42. What part of the country favored slavery?
43. Why do people use baking powder?
44. Describe some animals found in the North.
45. Who was the commander-in-chief of all the French armies?
46. Why is it possible to hear better by night than by day?
47. What labor saving machinery is used for farming?
48. What race of people fought with tomahawks?
49. From what dairy product is cheese made?
50. Name some product made from petroleum.

INTELLIGIBILITY LIST 19

1. Explain how brush bristles are fastened.
2. Which travels faster, a ferryboat or battleship?
3. Which is more expensive, cotton or silk?
4. In what year did Columbus discover America?
5. Name a man whose statue you have seen.
6. What is the longest river in the United States?
7. Which season do you prefer, winter or summer?
8. What house do caterpillars make for the winter?
9. Tell how Lincoln met his death.
10. Name two Hudson excursion boats.
11. What process makes peanuts digestible?
12. What canal connects the Hudson and Lake Erie?
13. How does a shrub differ from a tree in height?
14. How are we able to tell time by the sun-dial?
15. Tell by what means trains have been derailed.
16. Who is the cartoonist for the "Mutt and Jeff" series?
17. Explain what efficiency represents to you.
18. Tell what monument stands for liberty.
19. Why are we unable to see the bottom of a river?
20. What holiday is noted for its display of fireworks?
21. Why are grasshoppers colored green or brown?
22. Name some invention of the past war.
23. Which is more comfortable, a pullman or a day coach?
24. What instrument do we use to sharpen pencils?
25. At what time of the year do we use a thermometer mostly?
26. What happens to your door bell when you push the button?
27. Why is it safest to have steel cars in subways?
28. Name three things made in a bakery.
29. Tell what is the danger of getting acid on clothing.
30. How are pearl-oysters secured from the bed of the ocean?
31. How does asthma affect a person's health?
32. For what action is Paul Revere famous?
33. Why is lubricating oil smeared on a curved track?
34. Give reasons why rats should be destroyed.
35. At what time is the warmest part of the day?
36. Draw an outline of some common fish.
37. Why are news-stands placed in subway stations?
38. Tell how cities care for their aged.
39. Name the last novel you have read.
40. What vehicles convey people across the river?
41. What colors can you see in a soap bubble?
42. Describe the appearance of steam from a kettle.
43. What is the advantage in using a carpet sweeper?
44. What kind of storms do they have in the desert?
45. With what instrument does the submarine make observations?
46. What number comes before thirty-seven?
47. Describe the difference between the thistle and the goldenrod.
48. Tell me who is the chief executive of the nation.
49. Tell me a country where coffee is produced.
50. What birds act as messengers of the air?

INTELLIGIBILITY LIST 20

1. What magazine deals with short stories and jokes?
2. Name a game in which cards are used.
3. In olden times what use was made of indigo?
4. What race of people first settled in Canada?
5. State the advantage of the submarine cable.
6. Name the cab service you would call in an emergency.
7. What people wear turbans on their heads?
8. Explain why battleships are painted gray.
9. Explain how you travel to the Western Electric.
10. In what part of the country are snow-shoes used?
11. Why is it necessary for us to breathe fresh air?
12. What bird says, "Polly wants a cracker"?
13. From what substance are elastic bands made?
14. Do we export or import tea and rice?
15. How often do you use the telephone during the day?
16. Why do seaports become large cities?
17. Why do boys like to go barefooted?
18. Name some fruits that grow in New Jersey.
19. How many letters are there in your last name?
20. Why do some people wear reading glasses?
21. What is the name of the island on which the Statue of Liberty is situated?
22. How long does the sand take to sift through an hour-glass?
23. Explain why icebergs are dangerous to shipping.
24. Why do people wear bathing caps?
25. What is the woodpecker's method of securing food?
26. Explain the use of the electric washer.
27. What novel is concerned with the past war?
28. Why is there a nap on only one side of a carpet?
29. What rate of interest do banks pay?
30. How old must a man be before he can vote?
31. What is the easiest way to dissolve sugar?
32. Explain why trains are crowded during holidays?
33. What price do we pay for a quart of milk?
34. Why do people wear rings on their fingers?
35. Tell why factories have tall chimneys.
36. Why is it essential to have nurses in public schools?
37. What ornaments are worn by people of today?
38. Tell me the color of the suit you are wearing.
39. How are bright colors affected by the sunlight?
40. What kind of soil is best for farming?
41. Name a state where fine horses are raised.
42. Contrast the color of the blue-jay and the crow.
43. Write down the names of two inventors.
44. Who is benefited most by Daylight Saving?
45. Name some hot nourishing beverage.
46. How may barren soil be made fertile?
47. Name your favorite tooth paste.
48. Are the propellers of an aeroplane situated in front or rear?
49. At what time of the year do we have snow?
50. Why do squirrels gather nuts in the fall?

INTELLIGIBILITY LIST 21

1. Describe the climate in the northern states.
2. Tell me how a goat fights when angry.
3. Why are window shutters constructed from slats of wood?
4. Draw a triangle tangent to a circle.
5. Name a fruit which grows in California.
6. What causes numerous vessels to be wrecked?
7. Why do plants grow more rapidly in a greenhouse?
8. How is drinking water free from germs?
9. What two countries do the Great Lakes separate?
10. Name the Governor of the State of New York.
11. Draw an outline of North America.
12. Give the advantages of a house boat for the summer.
13. Explain why fish are dried for market.
14. What instrument guided Columbus on his voyages?
15. What mills make Holland picturesque?
16. Name the country in which Pompeii is situated.
17. How does a blackbird differ from a sparrow in size?
18. From what country do Panama hats come?
19. What are you doing with the hours after supper?
20. Is a swan's neck long or short?
21. Explain why tack hammers have claws.
22. In what city do French fashions originate?
23. Why do factories blow whistles at noon?
24. Print five letters of the alphabet.
25. In what country does tea grow in abundance?
26. What service does a ship's stoker perform?
27. What night school holds classes in drafting?
28. Name some hotels you would direct travelers to.
29. Why is it a bad policy to charge things?
30. How would you remove ink stains from your hands?
31. How many people are required to play chess?
32. Name the machine on which books are printed.
33. Which fruit do you prefer, apples or oranges?
34. Describe the consistency of molasses on a cold day.
35. Name the Scotchman's favorite cereal.
36. Draw two parallel lines.
37. In what places are dogs used for pulling sleds?
38. To what country does the ship called the "Rotterdam" belong?
39. What methods are used for transporting mail?
40. Name three cities in New York State.
41. Tell what Y. W. C. A. signifies to the public.
42. Who was general of the United States forces in France?
43. What kind of pipe did the Indians smoke?
44. Describe the effect of water upon fire.
45. What does a farmer do to improve his soil?
46. What canal separates North from South America?
47. Which do you prefer, roast beef or mutton?
48. Which is cheaper, a newspaper or the "Literary Digest"?
49. Explain why bears hide away for the winter.
50. What substance is used for painless dentistry?

INTELLIGIBILITY LIST 22

1. Where are earthquakes most frequent in our country?
2. What gritty substance does an ink eraser contain?
3. Why does the eagle terrify other birds?
4. Compare the size of a river and a brook.
5. Give the number of pecks in a bushel.
6. On what form of construction is an architect employed?
7. Describe the difference between a profession and a manual vocation.
8. What people are suffering for food today?
9. What climate is best suited for growing oranges?
10. In what part of the United States is there dry farming?
11. Which metal is more expensive, gold or silver?
12. What part of the milk do we use for butter?
13. Draw a diagram of the listening booth.
14. Why do balloons ascend when sand is cast out?
15. Why is mortar necessary for brick laying?
16. Name a place where public concerts are given.
17. Why are employees obliged to punch time-clocks?
18. What becomes of salt when mixed with water?
19. In what city is our state prison situated?
20. Give a beneficial use of the modern telephone.
21. What are the sources of profit from sheep ranching?
22. For what man was the State of Pennsylvania named?
23. For what reason do you take a camera on an outing?
24. Explain why we celebrate Decoration Day.
25. Which is the more expensive, silver or platinum?
26. Decoration Day comes at what time of the year?
27. Tell which is larger, a teaspoon or a tablespoon.
28. What is the color of the house in which the President lives?
29. In what season are palm beach suits worn?
30. Explain why subway tickets are chopped.
31. Name a food manufactured from grain.
32. Why should we eat fatty food in winter rather than summer?
33. What ocean borders the western coast of America?
34. How many things make one dozen?
35. What domestic bird is kept as a pet?
36. Tell why numbers are put on houses.
37. Name an ally of Germany in the past war.
38. On what occasion is it necessary to wear a bandage?
39. What kind of power is furnished by the Niagara Falls?
40. What is a public institution for delivery of mail called?
41. Which has a sweet taste, borax or table salt?
42. What small fish are preserved in oil?
43. After a thunderstorm what often appears in the sky?
44. What machine is utilized for shoveling dirt?
45. What musical instrument is most extensively used?
46. Explain why boys collect marbles.
47. Explain the chief business of a detective.
48. Where can articles of hammered brass be purchased?
49. With what substance does a dentist crown your teeth?
50. What time of the year is best for camping?

INTELLIGIBILITY LIST 23

1. Name a person that has donated to public charities.
2. At what time of the year do thunderstorms occur?
3. Mention a monopoly that exists today.
4. Tell about the event for which Galveston is noted.
5. Why do the railroads advocate an increase in fare?
6. How valuable are indications of approaching earthquakes?
7. Give a word that has one letter in it.
8. What becomes of ice that is exposed to the sun?
9. What nutrients are used in making macaroni?
10. What fashions of today would you like to change?
11. Why is the testing booth built sound proof?
12. What kind of work is accomplished in a kindergarten?
13. Why do aviators wear fur-lined coats?
14. Why do merchants display goods in their windows?
15. What danger is there from a space between a platform and a train?
16. What happens to your hair when brushed briskly?
17. What work is done by hand-writing experts?
18. Why is it well to avoid such places as penny arcades?
19. On which floor of the Western Electric do you work?
20. Why do people wear light clothing in summer?
21. Mention a poem and name its author.
22. Name some of the largest fishes.
23. For what purpose does the United States Government issue patents?
24. Why is stale bread better for us than fresh?
25. What musical instrument is played by blowing?
26. Which is better in business, longhand or shorthand?
27. What season of the year do you enjoy most?
28. Where do we deposit our money to insure its safety?
29. Who was commander of the American Expeditionary Forces?
30. Explain how the index of a book is a benefit.
31. In what part of the house is the garret built?
32. Why do we keep our glasses in a case when not in use?
33. Name the country in which Paris is situated.
34. Explain why varnish preserves paint.
35. What anæsthetic is administered to people before an operation?
36. Name a place in New York where there is a zoo.
37. Why do electric trains have third rails?
38. Name the chief uses for electricity.
39. Name two western states.
40. How long does it take to travel from New York to California?
41. What country perfected aerial telephony?
42. Name the window placed in a roof.
43. What is meant by the population of a city?
44. Of what special use is a daily newspaper?
45. Why are ferry boats built the same at both ends?
46. Mention the island where manila hemp is found.
47. What kind of trees are best for shade?
48. Why do firemen employ scaling ladders?
49. Name the sand hills along the seashore.
50. Which side of an equilateral triangle is longest?

INTELLIGIBILITY LIST 24

1. What is the difference between a local and express subway?
2. Why does corduroy make good trousers?
3. Describe the color of the olive tree.
4. To which state does Long Island belong?
5. Why does smoke go up chimneys?
6. What use do you make of your finger nails?
7. How did the United States raise money to carry on the war?
8. What group of islands is near the United States?
9. From what animals are mole skin coats manufactured?
10. Where is the Great Salt Lake situated?
11. Which is more invigorating, a walk or a drive?
12. Mention an implement which hunters use.
13. In what state is shoe manufacturing carried on extensively?
14. Name two states south of New York.
15. Describe the appearance of sharks' teeth.
16. Explain the difference between reveille and taps.
17. What animal with a large mouth lives in Florida?
18. Why does wall paper come in rolls?
19. Which is warmer, Canada or Central America?
20. What use do we make of a pair of scissors?
21. What company produces phonograph records?
22. Name a liquid for cleaning clothes.
23. Which is more sly, a fox or a bear?
24. Name one of the American battleships.
25. Name a dark object and a light object.
26. Give the name for dried grapes.
27. What benefits are derived from agricultural colleges?
28. Give a name of a Christmas story.
29. Describe the color on the inside of an oyster shell.
30. Why, in olden days, did castles have towers?
31. Name three large islands in the world.
32. How are steel structures held together?
33. Name some furniture made from mahogany.
34. Why are ticket men placed at the entrance to theatres?
35. Name an implement that is used for climbing.
36. Tell where birds build their nests.
37. When do you intend to come and visit me?
38. What kind of soil is found on the seashore?
39. Name an animal that lives in the water.
40. Why don't you take a trip around the world?
41. Give the abbreviation for one pound.
42. What do you think of a soldier who runs from the enemy?
43. Do owls sleep during the day or the night?
44. Which do you prefer to ride, a motorcycle or a bicycle?
45. From what part of the plant do we obtain tobacco?
46. Which is the greater offender, the briber or the bribed?
47. What do you generally see in a cream of wheat advertisement?
48. Tell what net weight means.
49. Tell the necessity of blotting paper.
50. How great is the humidity before a thunderstorm?

INTELLIGIBILITY LIST 25

1. What kind of relaxation do you indulge in after business?
2. Of what material is the chimney of a lamp made?
3. What causes boiling milk to overflow?
4. What Red Cross supplies were in readiness for the soldiers?
5. Mention a subject for a school debate.
6. What do we use to find the meaning of a word?
7. What letter do we add to "pencil" to form the plural?
8. Why are children happy when the snow falls?
9. Explain the utility of adding machines.
10. Which is the more strenuous, basketball or golf?
11. Tell why engines are painted black.
12. How many times a day do you clean your teeth?
13. What causes chapped hands in winter?
14. Name a plant that grows in water.
15. Explain how they wash the streets in summer.
16. Why do people exterminate mice from their homes?
17. Who must have the greater training, stenographer or typist?
18. Why can isin-glass be placed in stoves and furnaces?
19. Of what material are the stairs of this building composed?
20. Give the number of the department in which you work.
21. What happens to water when exposed to the sun?
22. Does a robin run, jump or hop?
23. What letters stand for District of Columbia?
24. What is a pleasant way to go to West Point?
25. Name a celebrated musician of the past.
26. Name a firm which manufactures victrolas.
27. What instrument is made for taking pictures?
28. Mention a state bordering on the Pacific Ocean.
29. Explain why a penknife is a necessity when camping.
30. What food products does Heinz prepare?
31. What is the rank of the lowest commissioned officer?
32. Why are chimneys built wide at the bottom?
33. Name the largest flowers you know?
34. What happens to the leaves of trees in autumn?
35. Name a state on the Atlantic coast.
36. How many quarts make a gallon?
37. How is gas conducted to city homes?
38. What does the name of Wall Street immediately suggest?
39. How many stars are there in the Great Dipper?
40. After what day of the week does Tuesday come?
41. What part of the potato plant do we use for food?
42. What is used in case an ink blot is made?
43. Of what material were the soldiers' uniforms made?
44. What cereals are served with sugar and cream?
45. What is the best weather for taking a photograph?
46. At what hour in the day does the afternoon begin?
47. On what occasion is it necessary to use a calendar?
48. Of what political party are you a member?
49. What large fish is known as a fish eater?
50. What makes the noise when a bag bursts?

INTELLIGIBILITY LIST 26

1. Why is it dangerous to put lead pencils in your mouth?
2. Why do employees dislike to punch the time-clock?
3. Mention an important city on the Hudson River.
4. Why is it wrong to throw banana peels on sidewalks?
5. Name a flower that is the color of an amethyst.
6. From what part of the plant do we get tea?
7. How could you reach the Palisades from Manhattan Island?
8. Why do bakeries wrap bread in wax paper?
9. What is the meaning of the expression, "Over the Top"?
10. Do you have green or white shades at your windows?
11. At what public places are money orders issued?
12. Tell why railways use the block system.
13. Tell how mountains differ from valleys.
14. Which exercise is more beneficial, walking or swimming?
15. In what country was Christopher Columbus born?
16. In what ways could you communicate with a person in Chicago?
17. When are the nights longer, in summer or in winter?
18. What mathematical symbol is used for minus?
19. Which do you prefer, gas light or electric light?
20. What kinds of timber are used for furniture?
21. Why are streets sprinkled in hot weather?
22. Name two engineers in the Transmission Branch.
23. Name your favorite out-door sport.
24. Do potatoes grow on trees or vines?
25. Why do we put perishable food in a refrigerator?
26. Tell how a bicycle can be steered.
27. Tell why it is important for people to have bank accounts.
28. Draw a diagram of a full moon.
29. What is meant by an amendment to the Constitution of the U. S.?
30. Tell what political party you uphold.
31. Why is it essential to have a good education?
32. Why are large signs put in front of stores?
33. Which is the higher building, the Woolworth or the Western Electric?
34. Do you consider the action of pirates just or unjust?
35. What advantage is there in using a safety razor?
36. Mention an American who has risen from the ranks.
37. What part of the country objected to slavery?
38. Tell what is meant by saying "Au Revoir."
39. Who played the last baseball game you attended?
40. What is the chief attraction of Greenwich Village?
41. Of what materials are glaciers chiefly composed?
42. Where would you hang wet clothing to dry?
43. Which is greater, the length or width of Manhattan?
44. What do ranchers use as bait to catch wolves?
45. Name the lake on which Chicago is situated.
46. Explain the convenience of a moving stairway.
47. Name the rough covering of a tree trunk.
48. Describe the feet of water birds.
49. What countries does the Suez Canal separate?
50. Name the country of which Paris is the capital.

INTELLIGIBILITY LIST 27

1. Name a summer resort on the seashore.
2. How many arms has a starfish?
3. Tell what people are fond of spaghetti.
4. Name a large city in Canada.
5. Write down your approximate weight and height.
6. Tell what trees are tapped for their syrup.
7. Why does hot air rise rapidly?
8. Name a famous woman of the White House.
9. Locate the business section of this city.
10. Of what use are claws to a cat when walking?
11. Comment on the benefits of labor unions.
12. What is the difference between a horizontal and vertical line?
13. Name the flower that grows on the hills of Scotland.
14. How does the Hudson Tube facilitate travel?
15. Name an animal that croaks and one that cackles.
16. Explain the method of popping popcorn.
17. Explain what is meant by a full moon.
18. How many propellers has an ocean liner?
19. Describe the appearance of a Scottish Highlander.
20. Name the tall grasses that grow in ponds.
21. Discuss the value of canteens to soldiers.
22. Why do so many immigrants come to America?
23. Of what material are some bathing suits made?
24. How many inches are there in two feet?
25. Discuss some application of nitroglycerine.
26. Which do you prefer, brown bread or white bread?
27. What insect stores away food for the winter?
28. What name is given to a person authorized to examine accounts?
29. Of what forces was General Foch in command?
30. Why do people ride on horseback in the park?
31. What railroad terminal is situated at Forty-Second Street?
32. What sounds does a mocking bird imitate?
33. Who was President of the country during the Civil War?
34. Briefly define the word militia.
35. What was the cause for inventing textile machinery?
36. Spell a word meaning the same as "divide."
37. Tell why it is necessary to have furniture in our houses.
38. Tell me how frequently you attend church.
39. Name the substance from which maple sugar is made.
40. Why is it necessary to take care of our eyes?
41. What city is noted for being devastated by a flood?
42. Why do animals in the north have thick fur?
43. Name three leading newspapers in New York.
44. Why are gold and platinum prized so highly?
45. In what way has the Panama Canal affected commerce?
46. What is the color of the gas flame used for cooking?
47. Why do people sleep at night instead of in the daytime?
48. Name the mayor of New York City.
49. How does ham differ from fresh pork?
50. Explain the reason for signal lights in a subway.

INTELLIGIBILITY LIST 28

1. Name five four-legged animals.
2. Which has three sides, a triangle or a square?
3. By what power are subway trains transported?
4. Why must we be careful of a sharp knife?
5. Explain how a snail carries his shell.
6. Name a country from which cheese is imported.
7. In what way is an illiterate person handicapped?
8. Through what process does water go to create steam?
9. Name an article used for unlocking doors.
10. Why is it necessary to have substantial chimneys on our houses?
11. Explain how Canal Street received its name.
12. Describe the flower which designates purity.
13. How many degrees are there in a right angle?
14. What carries our messages over the telephone?
15. Name an important scientific discovery.
16. Why is mountain climbing sometimes dangerous?
17. Name a state in which cotton grows.
18. Give two uses of iodine.
19. At what time of the year do we use holly?
20. By what means are we carried to the top of a high building?
21. How many times a week should we indulge in gymnastics?
22. Which amusement at Coney Island do you prefer?
23. Which do you prefer, moving stairways or an elevator system?
24. Why did the French Huguenots come to America?
25. What happens to water in a vessel over a fire?
26. Name the country from which the Italians come.
27. What transportation line goes under the Hudson River?
28. What country had the largest army before the war?
29. How is raw cotton changed into cloth?
30. Why is a variety of rabbit called a cotton-tail?
31. In what kind of houses did the pioneers live?
32. What happens when a water main breaks?
33. Name a magazine you have recently read.
34. Why does the law require tail-lights on automobiles?
35. Which is the more digestible, cooked or uncooked eggs?
36. Which is the higher part, baritone or tenor?
37. How often do you write letters to your friends?
38. For what reason is the war cross given?
39. Name two commodities exported from America.
40. How often does a century plant bloom?
41. What is the most common bird in the vicinity?
42. What city was first named New Amsterdam?
43. Name two large lakes in the United States.
44. What magazine gives the topics of the day in condensed form?
45. State some advantages of country life.
46. Name one of the best made automobiles.
47. What forms on fresh milk when it stands?
48. Why are the trains crowded before nine in the morning?
49. Name your favorite breakfast cereal.
50. Suggest a famous summer resort.

INTELLIGIBILITY LIST 29

1. How can a bouquet of flowers be kept fresh?
2. How does cold storage affect the taste of food?
3. For what luxury did the Indians trade corn?
4. What is the color of the cherry tree blossom?
5. In what war was Gettysburg a scene of hostility?
6. Write down the abbreviations for Pennsylvania and Maine.
7. Compare the size of the ocean and a lake.
8. Name an electrical invention which lightens work.
9. Name two uses for aeroplanes.
10. Who delivers the mail to our homes each day?
11. What kind of nest does a humming-bird build?
12. Which comes first in a storm, thunder or lightning?
13. Name a singing bird which is an American favorite.
14. What is the chief business carried on in Wall Street?
15. How would you describe a person who has no ambition to work?
16. Name one British colony.
17. What time in the morning do you leave for business?
18. Tell why Alaska is a valuable country.
19. State what is meant by kidnapping a child.
20. Explain what drives a compressed-air drill.
21. Explain why it is an advantage to be able to swim.
22. Why are the finger prints of every criminal taken?
23. Why do wild ducks spend the summer in the park?
24. How many wheels has a wheelbarrow?
25. What vocation is your main ambition?
26. In what place do you mail important packages?
27. Name some foods which are prepared in the oven.
28. What is the advantage of having telephone directories?
29. What causes the loud noise when a gun is fired?
30. Who presented America with the Statue of Liberty?
31. Tell why encyclopædias are necessary in school.
32. What material is used in the construction of a Japanese home?
33. Why do we wear heavy clothing in winter?
34. Why does smoke blow in a certain direction?
35. What article of food is called the staff of life?
36. What does the Statue of Liberty hold in her hand?
37. Mention a state bordering on the Atlantic Ocean.
38. At what time of the day do you feel most ambitious?
39. With what magazine are the Western Electric people supplied?
40. Name a game that can be played on the lawn.
41. How would you treat a burn to stop the pain?
42. In what year did the World War begin?
43. What is the most precious metal known?
44. Name a tree on which mistletoe is found.
45. How many years does biennial represent?
46. What instrument does a sailor use for signalling?
47. Tell what beverage is obtained from grapes.
48. Which is more densely inhabited, land or sea?
49. Of what benefit is the Fire Department to the city?
50. Why is it necessary to blow an automobile horn?

INTELLIGIBILITY LIST 30

1. What farm products are obtained from the country?
2. Name one of the large public markets.
3. Discuss the utility of a submarine chaser.
4. Why is there a gas meter installed in your home?
5. What is the advantage of having department numbers?
6. Name a food product that comes from Egypt.
7. Name a country famous for its clocks.
8. Why do birds fly south in winter?
9. Why are telephones placed in sound-proof booths?
10. Which is harder, hard coal or charcoal?
11. Tell when it is necessary to extract a tooth.
12. How can we tell that a dog loves his master?
13. Name a large insurance company in this city.
14. Why are sofa cushions arranged on a couch?
15. Why do we carry traveling bags when traveling?
16. What material would you use to start a fire?
17. Explain the preference for colored lamp shades.
18. Why are mirrors placed in drawing rooms?
19. Where do we find oil wells in the United States?
20. Write down the number of pounds in a ton.
21. Why is it uncomfortable to walk on cobbled streets?
22. Name the house in which chickens are kept.
23. What inventions have been announced by this company?
24. Tell what animals go in flocks.
25. What newspaper publishes a comic section?
26. Do we export or import cotton and tobacco?
27. Discuss the advantage of advertising an article.
28. What makes an electric fan revolve?
29. What swamp animal furnishes leather for bags?
30. What is necessary in the heading of a business letter?
31. Write down the name of our national anthem.
32. Why is it necessary for the city to employ garbage wagons?
33. How often does the circus come to town?
34. Give the color of the iris of your eye.
35. Which is the larger unit, a division or a company?
36. Why are hose wagons necessary to the fire-department?
37. Why do we open our windows in summer?
38. What happens to vapor when it rises from the earth?
39. What was the lighting system in olden times?
40. Name a flower which you like to smell.
41. Who is your favorite moving picture actress?
42. Why is it necessary to have floor walkers in department stores?
43. What kind of fuel do we burn in our fires?
44. Why is it necessary to water plants in a garden?
45. Name a shellfish used for food.
46. Name some important uses for gold.
47. What kind of wells supply wealth to Oklahoma?
48. Of what advantage are a pair of opera glasses?
49. How do banks make use of the money deposited with them?
50. What made Sergeant York famous in the war?

INTELLIGIBILITY LIST 31

1. Name a food that the cocoa bean produces.
2. What is necessary to improve the housing problem?
3. What large horned deer is found in Canada?
4. What occupations must be carried on in the cities?
5. From what wire is steel cable made?
6. What is the postage charge for special delivery?
7. Why do buildings have revolving doors in winter?
8. Of what value are Red Cross dogs during war?
9. Which is more stimulating to the system, water or coffee?
10. Why do we get a headache in a crowded room?
11. Where are pine trees found in abundance?
12. Name the cluster of lights suspended from the ceiling.
13. Draw a circle in a square within a triangle.
14. Tell me how many hours there are in one week.
15. Name an island in the Pacific Ocean.
16. Tell the story of the Boston Tea Party.
17. For what purpose are amusement piers built?
18. Why do you take photographs of places you visit?
19. How many pictures did you take last summer?
20. Name a vegetable that grows on a cob.
21. How are automobile tires prevented from skidding?
22. Which is the better lighting system, direct or reflected?
23. Explain why a cat has padded paws.
24. What do the thirteen stripes in our flag represent?
25. Why is it necessary for people to wear eye-glasses?
26. Name the last day of the week.
27. Who receives the larger salary, president or office boy?
28. What machine is used for printing paper?
29. What meat do you prefer for a camp fire roast?
30. Name two transcontinental railroads.
31. What winds usually bring rain?
32. Why do boys raise pigeons for market?
33. How far did you walk this morning?
34. Did the early Americans cook in fireplaces or stoves?
35. What do you find on doors of a business office?
36. Why is the camel a valuable animal in the desert?
37. Name a building which is noted for its architecture.
38. With what day is Santa Claus associated?
39. During what war were gas masks used?
40. Where do many of the commuters to this city live?
41. Describe the kind of soil in your community.
42. Name a public park of New Jersey.
43. Which kind of berries do you like best?
44. What is the most important use of the ocean?
45. Why do ships blow their horns in a fog?
46. What is the most expensive stone mined?
47. Why is it good to carry a gun on a camping trip?
48. Which is more economical, gas or electricity?
49. What president's picture is engraved on a penny?
50. What valuable fur-bearing animals are natives of Alaska?

INTELLIGIBILITY LIST 32

1. What is the outside covering of a tree called?
2. Compare the sailor's uniform in summer and winter.
3. Do you select jewels for their beauty or their value?
4. Tell from what grain flour is made.
5. Name the shell-fish from which pearls are obtained.
6. Describe the manner of arranging asparagus for market.
7. When can we see the stars, during the day or night?
8. What part of a theatre is called the balcony?
9. Which is the better dessert, short-cake or buns?
10. What name is given to the man who shoes horses?
11. What two oceans are of most importance to us?
12. What is the chief cargo of boats on the Mississippi?
13. How is water obtained from a well in the country?
14. Why is the fireless cooker a convenience in the home?
15. Explain how sea-gulls secure their food.
16. Give the use of one of the five senses.
17. How do you fasten a pair of glasses on your nose?
18. From what material are automobile tires made?
19. What is meant by a chain grocery store?
20. What would happen if you did not have an alarm clock?
21. Name the material from which awnings are made.
22. Explain the necessity for a drawbridge across a river.
23. Write the summation "five plus seven."
24. Of what advantage are telephone instruments to the public?
25. Why is Niagara Falls visited by the public?
26. At what season of the year should seeds be planted?
27. What is the quickest way to add a column of figures?
28. Which is more valuable, silver or tin?
29. Tell what a bird's topknot consists of.
30. Where do we find some very large trees?
31. Why was a bridle-path made on Riverside Drive?
32. What mail is included in second-class matter?
33. What is the color of the eyes of a white bunnie?
34. Name the man who presides over a school.
35. What are beans that climb poles called?
36. Name the deer that the Eskimos have domesticated.
37. Describe some of the advantages of living in the city.
38. Why is the pitchfork a convenient garden implement?
39. What materials are used for shelter in our country?
40. What kind of rubber heels are used by the public?
41. In what direction do the roots of a plant grow?
42. How much money do you spend on soft drinks?
43. Name a product of a tropical country.
44. Name some of the New England states.
45. How many eggs make a dozen?
46. Name some authors of English literature.
47. What letters go to the dead letter office?
48. In what climate is cane sugar raised?
49. At what time of the year should we expect blizzards?
50. Name a summer resort on the Jersey coast.

INTELLIGIBILITY LIST 33

1. What substance does a jelly fish resemble?
2. Name the country to which India belongs.
3. Distinguish between a wild flower and a cultivated flower.
4. Give a description of a shadow at noontime.
5. Why do bathing resorts have life-saving stations?
6. Give an important use of the tugboat.
7. Name a food that contains starch.
8. Name the founder of the League of Nations.
9. What places in the West are noted for their scenery?
10. To what part of the United States are missionaries sent?
11. For what purpose is a moving picture machine used?
12. Name an article made of wood pulp.
13. Tell why fish are valuable to men.
14. Why do communities maintain art galleries?
15. Into what river do the Great Lakes empty?
16. What kind of wire is used for conducting electricity?
17. Name a large seaport in Massachusetts.
18. Where do people place their valuables for safe keeping?
19. Describe the consistency of cement after drying.
20. Tell what animals go in herds.
21. Name a firm that manufactures milk chocolate.
22. Name a famous tower at Madison Square.
23. Why do sea-gulls follow a ship?
24. Name the place in a desert where water is found.
25. What is supplanting horses in the Fire Department?
26. What would you expect to see in a good aquarium?
27. What is the difference between a gas engine and a steam-engine?
28. What are two vegetables whose roots we use for food?
29. Name a game which is played with racquets?
30. Give the number of smoke-stacks on the Mauretania.
31. What effect does the gulf stream have on the climate of New York?
32. Name the building in which our currency is made.
33. What is the accepted theory about the moon's surface?
34. Name the first month in the year.
35. Why are docks built along the Hudson?
36. By what means can the police detect a robber?
37. Name a fruit that has one pit.
38. Name a successful soap dye.
39. How fast can a modern locomotive travel?
40. What insect destroys our clothing in summer?
41. Tell me which musical instruments you can play.
42. How many elevated railways are there in New York City.
43. In what country did the suffrage movement originate?
44. Why does everything become colder at night?
45. From what country did we import negro slaves?
46. Is steam heavier or lighter than air?
47. How does a palm leaf differ from a maple leaf?
48. In what part of the body are the vocal cords?
49. What help can a ship obtain from a lighthouse?
50. What happens when an earthquake shakes a city?

INTELLIGIBILITY LIST 34

1. Why are automobiles required to carry license plates?
2. Compare a watch and a clock as to size.
3. What materials are used for paving streets?
4. What bridge extends from New York to Brooklyn?
5. Why is the sparrow considered an undesirable bird?
6. In what way do we make use of a barometer?
7. Where did an earthquake prove fatal to human life?
8. Describe the peculiarity of a swallow's tail.
9. In what way can scratched furniture be renovated?
10. How many stars were there in the first American Flag?
11. In what war have France and England been involved?
12. What time of the year do boys fly kites?
13. Why are fresh eggs necessary for invalids?
14. What crops are raised exclusively in the South?
15. Is a launch or a canoe more dangerous?
16. Why are seals regarded as valuable animals?
17. Explain why individuals have different names.
18. Why are ambassadors sent to foreign lands?
19. Name the favorite food of the Italian.
20. For what purpose is a window pole used?
21. Why do fishermen spread their nets on the sand?
22. What enables a grasshopper to hop a long distance?
23. What commercial article is made from a palm leaf?
24. What substances do you add to coffee before drinking?
25. Of what age are the children who attend kindergarten?
26. Name two ways in which glass is utilized.
27. Twenty-five cents is what part of one dollar?
28. Tell what is necessary to wear in a rainstorm.
29. Give the name of a pleasure boat.
30. What kind of fur do animals in the North have?
31. Mention some Indian arts and crafts.
32. Name a country of South America.
33. For what purposes is anthracite coal used?
34. Describe the kangaroo's method of locomotion.
35. How are ladies' hats trimmed in summer?
36. How many Liberty Bonds do you own?
37. What domestic animal would you choose for a pet?
38. Explain the necessity for a looking glass.
39. What do we call the man who blows a bugle?
40. Suggest an idea for the housing proposition of New York.
41. Name a military school situated on the Hudson.
42. Why is farm life more healthful than business life?
43. In what part of the country are ranches found?
44. What is the name of your favorite fairy tale?
45. Why do men wear cuff buttons?
46. Why is tooth paste put up in tubes?
47. What impresses you most when you attend a circus?
48. How often do we hold our presidential election?
49. By what means does a horse protect itself?
50. Explain why typewriters are important in business.

INTELLIGIBILITY LIST '35

1. Give a common slang expression.
2. When are white caps visible on a lake?
3. Give a reason why it is wrong to kill birds.
4. Explain why a piano lamp is on a tall stand.
5. Explain the advantage of a checking account.
6. Tell what happens to a punctured tire.
7. What can you say of the nutritive value of peanuts?
8. For what purpose is the thermos bottle used?
9. Why is it dangerous to pour gasoline on a coal fire?
10. Name two ways of using gasoline.
11. How may electricity be used to diminish work?
12. Why are there so many tall buildings in this city?
13. Explain how whipped cream is prepared.
14. How much rain falls in the desert?
15. What is the advantage of storing coal in cellars?
16. How many blades has a pair of scissors?
17. Name a current magazine which you like to read.
18. At what time is the coldest part of the day?
19. Why are casters used under heavy furniture?
20. Name the largest jewelry store in the city.
21. How does a small object appear through a magnifying glass?
22. Give an example of a climbing plant.
23. Why is tennis considered a first-class sport?
24. On what occasion is it necessary to use an umbrella?
25. Why is silk preferred to cotton for umbrellas?
26. Draw a circle within a square.
27. What advantage is derived from studying geography?
28. Tell me where you can purchase piano wire.
29. How many hands has an alarm clock?
30. Why do we like ice cream in hot weather?
31. What is the difference between a sweet and white potato?
32. Explain the utility of coat hangers.
33. Name the memorial arch in Washington Square.
34. What is meant by the expression "See America First"?
35. Explain when the services of a lawyer are solicited.
36. What happens to raindrops that fall on the ground?
37. What month do holly and mistletoe suggest?
38. Give a brief definition of an island.
39. Tell of what importance maps are to sailors.
40. What machine is used for weighing freight?
41. How does a wheelbarrow assist a gardener in his work?
42. Why is it necessary to have head lights on automobiles?
43. Why are policemen stationed at street corners?
44. Why is it important to register valuable letters?
45. What fish spouts streams of water?
46. What is the name given to the executive of a city?
47. Tell how a dog shows his happiness.
48. What instruments are used to lock doors?
49. Where is salt found in the United States?
50. Contrast the daisy and black-eyed susan.

INTELLIGIBILITY LIST 36

1. Explain the difference between a subway and an elevator railway.
2. Which is more refreshing, skating or walking?
3. Name a country from which tea is imported.
4. What time do you expect to be home this evening?
5. Were Washington's parents humble or aristocratic?
6. What would happen if the food distribution ceased for a week?
7. Name two uses for ice.
8. What exploration made Peary famous?
9. After whom was Van Cortlandt Park named?
10. Why were radio watches important during the war?
11. Explain the difference between a medical and osteopathic treatment.
12. How does a cat show her anger when teased?
13. What is the advantage of shopping in a department store?
14. Tell what harm is done by tornadoes.
15. What is the boundary line between Texas and Mexico?
16. Name an insect that stings and makes honey.
17. What is the purpose of compulsory military training?
18. Explain what is meant by a circulating library.
19. Name some furniture made from oak.
20. On what hemisphere is North America situated?
21. Name a famous Confederate general.
22. What blows about the dandelion's seeds?
23. Why are immigrants subject to a health test?
24. By what telephone extension can you be reached in this building?
25. Why is it best to use a thimble when sewing?
26. At what place do we find the source of a river?
27. How did the ancients grind grain for meal?
28. Give a cause of the Civil War.
29. Name a city where flour is milled.
30. Where is anthracite coal mined in this country?
31. Explain why you take the shuttle at Times Square.
32. How many people form a quartet?
33. Compare the size of the earth and the sun.
34. Describe how the earthworm moves about.
35. What part does a clown take in a circus?
36. On what river is the city of Washington situated?
37. How many hours a night do you sleep?
38. How many months are left in this year?
39. Name the country to which Alaska belongs.
40. Discuss the use of fire-escapes on buildings.
41. Name a president of the United States who was assassinated.
42. In what way are subways beneficial to New York?
43. What name is given to a body of salt water?
44. What games are played in Van Cortlandt Park in summer?
45. How do police catch criminals by finger prints?
46. Why are mills and factories built on rivers?
47. Name the two governing bodies of the United States.
48. What sort of business is handled by an auctioneer?
49. What help are public playgrounds to children?
50. Mention some lawyers you know.

INTELLIGIBILITY LIST 37

1. Name the Quaker city in Pennsylvania.
2. How were houses illuminated before petroleum was discovered?
3. What is the difference between a soprano and contralto voice?
4. Why are pieces of rubber inserted in a horse's shoe?
5. To whom is the power to veto bills given?
6. What do we call a man who builds houses?
7. Why are wrist watches preferred to the old style?
8. What kind of oil is used in an automobile engine?
9. What kind of flower is the American Beauty?
10. Tell what use we make of our nostrils.
11. Name the state of which Albany is the capital.
12. Why is red lead applied to iron before painting?
13. Mention a foreign language that is familiar to you.
14. Tell how it is possible for the blind to read.
15. Whom do you consider the greatest military genius?
16. Why is it that deep sea fish do not have eyes?
17. What brand of pork and beans is highly advertised?
18. Why do autoists carry spare tires?
19. Why should the wire of an electric circuit be insulated?
20. What is the name of the person who made our first flag?
21. Which are warmer, wool or cotton sweaters?
22. How do cast iron articles receive their shape?
23. Why are people in tropical countries lazy?
24. Name a university in New York City.
25. What is the easiest way to sharpen a pencil?
26. Name something that makes you sneeze.
27. What event is commemorated by a holiday in July?
28. What work is done by a finger print expert?
29. Name a wild animal of Africa.
30. What occupations must be carried on in the country?
31. Explain what is meant by standard time.
32. What chimes are rung on Christmas Eve?
33. Name a firm which manufactures typewriters.
34. Explain what destruction may be caused by cyclones.
35. For what purpose is the anchor of a ship used?
36. In what way does it pay to advertise in magazines?
37. How can you remove a pencil mark from paper?
38. What happens when a man drinks liquor?
39. What is meant by canvassing the country for money?
40. What is the color of the uniforms worn by sailors?
41. Which has the greater nourishment, white or brown bread?
42. With what nation did we fight the Revolutionary War?
43. Name a fruit that has no seeds.
44. For what purpose are sugar beets raised?
45. What name is given to the man who measures ground?
46. What machine is used for cutting wood?
47. Compare the size of a watermelon and a cantaloupe.
48. Of what value are carrier pigeons during war?
49. Why is Manhattan Island so valuable today?
50. Name an oil that is procured from the earth.

INTELLIGIBILITY LIST 38

1. Give the sum of four and six.
2. Who are the more progressive, the ignorant or educated?
3. What city is noted for its manufacturing industries?
4. Who picks the cotton in the Southern fields?
5. How can we tell a true friend from a false one?
6. Why do birds that fish in swamps have long legs?
7. Name the smallest state in the Union.
8. Suggest a drink to relieve thirst.
9. Explain why fire-escapes are constructed of iron.
10. Why is it necessary to put corks in bottles?
11. Name the state in which Albany is located.
12. Name a boat that carries a search-light.
13. At what time of the year do we skate on the ice?
14. Why are jungle animals spotted and striped?
15. How is street traffic regulated on Fifth Avenue?
16. Name a firm that manufactures fine silverware.
17. What use do we make of carrier pigeons?
18. Explain the necessity of a Weed chain.
19. Mention some useful article made from bone.
20. How many cents make a dollar?
21. What do people learn from traveling in Europe?
22. What wind brings rain in New York?
23. Name the plant from which cocoa is made.
24. Why do poor peasants wear wooden shoes?
25. What substitute is used in the place of potatoes?
26. What instrument do we use to draw circles?
27. Tell why copper wire is valuable.
28. What colors are most becoming to a blonde?
29. What company has a manufacturing plant at Hawthorne?
30. What does the abbreviation M.D. stand for?
31. Give the name of an insect that flies and bites.
32. What kind of building stone have you seen?
33. Mention some materials from which newspaper is made.
34. How much sleep is required for a person over twenty-one?
35. What process makes brown sugar white?
36. Why are rewards generally offered for lost articles?
37. Name some things that farmers do.
38. What article of food is secured from a maple tree?
39. Why is a lightning beetle visible at night?
40. When did you last find a four leaf clover?
41. How many letters are contained in the alphabet?
42. What does it mean to draft men for an army?
43. Who are the more economical, the poor or the wealthy?
44. Name a delicious pie made from pumpkins.
45. Name a water animal that can learn tricks.
46. Mention a famous opera singer.
47. What people build houses of snow?
48. Which is colder, the northern or southern part of New York?
49. Name a flower which has thorns on its stem.
50. Name a college where summer school is conducted.

INTELLIGIBILITY LIST 39

1. Name some countries from which immigrants come.
2. Describe the sounds made by a dove.
3. What time of the day do we find dew on the grass?
4. What precious stone has a brilliant green color?
5. Why are mountains of use as summer resorts?
6. Name a city noted for its skyscrapers.
7. At what park are the New York Zoological Gardens?
8. Name some forts located near this city.
9. At what time of the morning do you eat breakfast?
10. By what means does a dog protect itself?
11. What is the approximate width of the Hudson River?
12. Give the color of iron rust.
13. How many fingers have you on both hands?
14. Why does the sun appear larger than the stars?
15. Explain the utility of opaque shades at our windows.
16. Where do most birds build their nests?
17. Name some commodity which we import from Japan.
18. Why is sugar added to apples before they are baked?
19. Name a beautiful bird that spreads its tail.
20. By what means is a motor boat propelled?
21. Explain why the snapping turtle receives its name.
22. Name the daily paper which you read.
23. Name a game that is played on horseback.
24. Write down the number of inches in five feet.
25. What toys are enjoyed by little girls?
26. Why are automobiles used in preference to horses?
27. Why is "Exit" painted in red over doorways?
28. Name the substance from which cardboard is manufactured.
29. What water system supplies New York with water?
30. Name two popular plays of the season.
31. How many inches are there in two feet?
32. Name the insect from which we get honey.
33. What is the basis for any higher profession?
34. Name the shortest month in the year.
35. At what time of the year do you get Spring fever?
36. What political party is in power at present?
37. Name the support from which a flag flies.
38. What is the number of the public school you attended?
39. Tell what people are very fond of rice.
40. What is the comparative size of the moon and earth?
41. Give reasons why flies should be destroyed.
42. Do tomatoes grow on vines or bushes?
43. What insects infest swampy regions?
44. What is the advantage of a waterproof coat?
45. Why can one man sell when another can't?
46. At what time is the sun highest in the heavens?
47. Why does a hare travel faster than a tortoise?
48. Why do people sometimes fall on the ice?
49. What wars have the Americans fought against the English?
50. Name one of the world's greatest sportsmen.

INTELLIGIBILITY LIST 40

1. What can you say about unemployment in the country at present?
2. What is meant by the boiling point of a liquid?
3. What are the principal parts of a telephone set?
4. Name the general whose tomb is on Riverside Drive.
5. Name a water-bird that has a long neck.
6. What stringed instrument is the easiest to play?
7. What kind of automobile are you able to drive?
8. Why do ships carry lifeboats?
9. Which is the heavier liquid, mercury or water?
10. Why is baking powder used in cake?
11. Why is it safest to sit quiet in a canoe?
12. For what reason are fireworks not allowed in this city?
13. Of what material do the Eskimos build their huts?
14. Give the reason for bats flying at night.
15. Name the seeds of a pine tree.
16. What tool is used to get rid of weeds?
17. What great statesman said "Lafayette, we are here"?
18. What kind of shoes are worn by the Indians?
19. Compare the speeds of a bicycle and automobile.
20. What is it your main ambition to become?
21. Explain how we can distinguish between a ferryboat and a tugboat.
22. How many points has the star in our flag?
23. Tell me where you have seen a natural waterfall.
24. Compare the attractions of the country and the city.
25. Describe the climate in which palms grow.
26. What do we call a man who draws plans for buildings?
27. Who established the five and ten cent stores?
28. How many angles are there in every triangle?
29. What materials are used for building houses?
30. Tell what is made out of willow for market.
31. After what famous man was Pennsylvania named?
32. How do you manage to wake up in time in the morning?
33. Mention a game which is played with cards.
34. Draw a vertical section of an apple.
35. Name the man who constructs a bridge.
36. State two uses for steam.
37. How did the discovery of gold affect the West?
38. What people take part in minstrel shows?
39. Which is easier to write with, a pen or pencil?
40. In what section of the city are the slums found?
41. How often did you go swimming last year?
42. What electrical appliances are installed in your home?
43. Mention two dairy products.
44. What is the purpose of a life insurance policy?
45. Why are advertisements posted in subway trains?
46. Of what value are opera-glasses to an observer at the theatre?
47. For what discovery is Benjamin Franklin famous?
48. What happens to a ball when thrown in the air?
49. Of what detriment are movies to school children?
50. Tell the favorite fruit of the South.

INTELLIGIBILITY LIST 41

1. Why does a turtle draw his head under his shell?
2. Name a tree that stays green all winter.
3. What part of the plant is the cauliflower we eat?
4. What month of the year are roses in bloom?
5. Why is it necessary for a ship to be taken to dry-dock?
6. Mention something that can be found at the bottom of the sea.
7. Discuss the services performed by a judge.
8. What part of the beet grows above the ground?
9. In what large city do we find stock-yards?
10. What vehicle is used for conveyance in the country?
11. Which is safer to ride in, a canoe or rowboat?
12. Name the most reliable goldsmith.
13. How is Thanksgiving Day celebrated in America?
14. Name some toys that amuse boys.
15. What is meant by a monopoly of the fur trade?
16. What are some of our best singing birds?
17. Why should rooms be fumigated after a contagious disease?
18. What immediate treatment should be given to a snake bite?
19. What tribe of people roam through the country?
20. What is best to do in case of fire?
21. During what war was Fort George used?
22. How may we become familiar with the best music?
23. What kinds of cases are treated in a hospital?
24. At what time of the year do birds come from the South?
25. Name a fine material that comes from China.
26. Compare the size of the Grand Central and the Post Office.
27. Explain why locomotives have large head-lights.
28. What part of a pail is called the bail?
29. What causes warm air to rise?
30. Tell what is used for explosives in mines.
31. What rate of interest do your Liberty Bonds pay?
32. How many days are there in a leap year?
33. Tell what the curfew bell signifies.
34. How would you invest money if you had plenty of it?
35. What is fast competing with automobiles and trains?
36. Name two of your favorite desserts.
37. Estimate the cost of a good typewriter.
38. Mention a body of water near Manhattan.
39. What paper do you prefer to read in the morning?
40. To which side of the road must traffic keep in this country?
41. From what nut is peanut butter made?
42. Name the bug that eats potato vines.
43. How are army officers distinguished from privates?
44. Why was a tax levied on theatre tickets?
45. What do three gold balls over a shop represent?
46. What kind of workmen wear overalls?
47. Name a country where people worship idols.
48. What artificial light is provided in your home?
49. How many sides has an equilateral triangle?
50. Tell the difference between a telescope and a microscope.

INTELLIGIBILITY LIST 42

1. What precious stones are used for ornaments?
2. What sport do men enjoy during the autumn season?
3. Name the animal into which the tadpole develops.
4. What kind of work is done by an employment bureau?
5. Of what is the greater share of watermelon composed?
6. What educational courses do you take in the evening?
7. What are two ways of ascending a tall building?
8. How many boroughs are there in New York City?
9. Where was the last World's Exposition held?
10. In what way are we annoyed by mosquito bites?
11. What month is affected by leap year?
12. What views do you most enjoy in this city?
13. Give the color of olive oil.
14. What materials are used to freeze ice cream?
15. What kind of shoes do you wear tramping?
16. How many senators are there in the United States Senate?
17. Explain why jewels are prized by people.
18. Why is it possible for rats to carry germs?
19. Explain the necessity for dredging rivers.
20. How does water make the pebbles round?
21. At what time of the year do chrysanthemums bloom?
22. Why is it impossible for fish to live in the Dead Sea?
23. From what continent does most of the coffee come?
24. How would you introduce an acquaintance to one of your friends?
25. Which is cheaper, freight transportation by canal or train?
26. In what country do the people dress in fur?
27. Who was the first man to cross the Atlantic by air?
28. In what part of the sky is the North Star located?
29. Name a railroad which runs into New York City.
30. Write down the country where diamonds are mined.
31. Name the state of which Trenton is the capital.
32. Tell the color of the holly berry.
33. Which is more expensive, a Pierce Arrow or an Overland car?
34. Name a popular southern melody.
35. Give a reason for using ink for writing.
36. What part of the United States does the Atlantic border?
37. Name the tree from which we get acorns.
38. What happens to a ball when thrown against the ground?
39. What part of a plant grows in the soil?
40. Name a machine used for getting calculations.
41. By what means is coal put into cellars?
42. In what famous mountains are the summer resorts of Europe?
43. Where are the best botanical gardens in this city?
44. Name two animals that climb.
45. Name three beneficial sports.
46. Name an old section of this city.
47. Name some products of mills and factories.
48. When does the baseball season begin and end?
49. Give the color and length of the earthworm.
50. What soap do you prefer for daily use?

INTELLIGIBILITY LIST 43

1. What avenue separates the East and West of New York?
2. Which is more strenuous, baseball or fishing?
3. What do the stars represent in the American Flag?
4. What is the name of one of the largest planets?
5. Name the first vegetables of spring.
6. Who composed a large and popular dictionary?
7. What is the title given a person who paints pictures?
8. Who are the famous brothers who make cough-drops?
9. What berry is found in swampy grounds?
10. What perils threaten ships in northern waters?
11. Explain why woodwork is freshly painted in the spring.
12. How long have you been employed by the Western Electric?
13. How is wall paper attached to the wall?
14. Explain why dogs should wear collars.
15. At what time is dew visible on the grass?
16. How many square feet are there in a square yard?
17. At what temperature does fresh water freeze?
18. What people have great faith in prophecy?
19. What hard covering envelops the turtle's body?
20. Name some dairies that supply milk for New York.
21. Name a health resort in New York State.
22. State the number of people in your family.
23. Name some department stores in this city.
24. Where are weather vanes found mostly?
25. Give the reason why horses are shod.
26. What kind of butter is made from peanuts?
27. What creature comes from a butterfly's eggs?
28. Why are watering troughs found throughout the city?
29. What is New York City's most fashionable street?
30. How long can camels go without water?
31. Why is it light during the day and dark at night?
32. Name a race that does artistic work.
33. What ways are there for traveling through the United States?
34. From what country was our Egyptian Obelisk brought?
35. By what method are lobsters caught for market?
36. Do you prefer to read prose or poetry?
37. Where do the materials for making our clothing come from?
38. Why are rubbers worn in rainy weather?
39. Name a poisonous wild plant.
40. What famous construction surrounds a part of China?
41. Name the principal food of the Indians.
42. Explain why quick-sands are a source of danger.
43. Give some reasons for advocating woman suffrage.
44. When do we have more electrical storms, in summer or winter?
45. Why is aluminum employed for cooking utensils?
46. How many pecks are there in two bushels?
47. What makes New York the commercial center of North America?
48. What tree bears nuts in the fall?
49. Why is plowing done by steam instead of horses?
50. How was cooking carried on before stoves were invented?

INTELLIGIBILITY LIST 44

1. For what purpose is an electric meter used?
2. Explain the cause of day and night.
3. What material was knit into garments for the soldiers?
4. Name the grain from which flour is made.
5. What caused the scarcity of sugar during the war?
6. Explain why the nation is interested in educating children.
7. What do we put in our refrigerators to keep our food cool?
8. What use is made of bituminous coal?
9. How is the stage of a modern theatre lighted?
10. Tell why windows are made of glass.
11. Why is it necessary for firemen to carry hatchets?
12. Explain how a cable car is propelled.
13. Describe the service performed by a bugler.
14. Give a reason for building the subway.
15. What is the best kind of weather for excursions?
16. How many colors are visible in a diamond?
17. Why is yeast used in baking bread?
18. Explain the necessity for dredging New York harbor.
19. Name a large hospital in this city.
20. Why is the lobster's body incased in a shell?
21. Tell me the consonants in your first name.
22. What substance is used to attach photographs in books?
23. Why is an axe handle made from hickory wood?
24. What fire extinguisher is placed in large buildings?
25. Explain where strawberries are grown in winter.
26. How can we tell to which country ships belong?
27. Name the four seasons of the year.
28. Name a king and queen of England.
29. Which are better for securing iron, nails or screws?
30. What company manufactures rubber heels for shoes?
31. Name three races of people in New York.
32. What three breakfast cereals have nutritive value?
33. What large bir dcatches mice at night?
34. What objects do you expect to see in a museum?
35. What is the quickest way to crack nuts?
36. What is meant by a school or shoal of fish?
37. Explain the difference between a sailor and a soldier.
38. What kind of food does the ocean furnish?
39. How many years are required for high school?
40. What is the shape of the home of the American Indians?
41. Why is fertile soil very valuable?
42. Why are many streets closed for play streets?
43. Name a large Atlantic steamer.
44. Name some valuable investment you have made.
45. Who have the privilege of voting in this state?
46. Why is it important for a train to have a whistle?
47. What southern state is popular in the winter?
48. Give our most important use for silver.
49. Why are drinking cups provided in public places?
50. Where did you go on your last automobile ride?

INTELLIGIBILITY LIST 45

1. Name two countries of North America.
2. Explain the reason for feeding horses grain.
3. What is the advantage of swimming in salt water?
4. Which is easier for cleaning, hot or cold water?
5. Name a mineral that is mined in the United States.
6. What man donated libraries through the country?
7. Explain why all countries have a standing army.
8. How does an object appear when viewed through a microscope?
9. Which is noticed first, the steam or the whistle?
10. Of what material do Eskimos make their clothing?
11. Of what material are piano keys made?
12. What places should a visitor see in New York City?
13. Which travels faster, a telegram or a telephone message?
14. How many hours do our employees work on Saturday?
15. Explain the difference between a sailor and a marine.
16. How do people protect themselves against a storm in a desert?
17. Why are dogs always with flocks of sheep?
18. Give five names of boys.
19. Why is it essential to have plenty of sunshine in our rooms?
20. Name a large city in California
21. Describe the climate of the southern states.
22. Where are Dodge and Ford cars made?
23. In what game are baseball bats used?
24. How many sides has a square?
25. Where will plants thrive, in fertile or barren soil?
26. Explain the necessity of a stairway in a house.
27. Name some ingredients in ice cream.
28. Since when has the cost of living been so high?
29. Who can do the heavier work, a man or a woman?
30. Of what benefit were aeroplanes during the last war?
31. In what way does an aeroplane resemble a bird?
32. What happens to our breath on a cold day?
33. Why are flowers raised in large quantities for market?
34. When does an ostrich bury its head in the sand?
35. Are the flowers of the tropics brilliant or pale in color?
36. How does an information bureau assist a stranger?
37. From what wood is mahogany furniture made?
38. What is the highest building in this city?
39. Tell where Bunker Hill monument is located.
40. Tell from what plant linen is obtained.
41. What are some games played by boys in the spring?
42. Mention an animal which we use for food.
43. Why is it necessary to have policemen in New York City?
44. Describe the appearance of storm clouds.
45. Describe the color of autumn leaves.
46. To what country did the Imperator formerly belong?
47. What is the color of the gas flame used for illumination?
48. What kind of bird builds a hanging nest?
49. Describe the appearance of snowflakes.
50. Name a nearby city in which there is a shipyard.

INTELLIGIBILITY LIST 46

1. Why is a license required of all dealers?
2. Which is the heavier substance, iron or water?
3. Tell what direction Denver is from New York.
4. Name the food administrator during the past war.
5. Why do people grow flowers in window-boxes?
6. State the author's name of "Uncle Tom's Cabin."
7. What is the object of cold storage concerns?
8. From what store do you prefer to purchase your clothing?
9. State what is meant by irrigating a garden.
10. Why do Indians make flour from acorns?
11. Estimate the cost of your last sickness.
12. Why are crackers packed in a tin box?
13. Explain why intelligibility tests are taken.
14. Mention a man who was prominent in history.
15. Name a boy's college in New Jersey.
16. Where was gold discovered in the United States?
17. Name an animal from which we obtain leather.
18. How much intelligence is necessary for matching pennies?
19. How do the sun's rays affect a burning glass?
20. Compare the leaves of the pine and oak.
21. What kind of heating system do you have in your home?
22. Why are gas logs employed in preference to wood?
23. What color is used for a danger signal?
24. For what reason are pleasure boats constructed?
25. What utensils are used for cooking outdoors?
26. What useful work does a cobbler perform?
27. For what purpose are the trunks of trees whitewashed?
28. What company makes Uneeda Biscuits?
29. Why was barbed wire so treacherous during the war?
30. Why are busses used on Fifth Avenue?
31. What fruit do you prefer with cornflakes for breakfast?
32. In what manner did the Indians travel over the water?
33. Which plant is larger, a tree or a shrub?
34. What insect's eggs produce caterpillars?
35. Tell how Central Park received its name.
36. How is an injured person removed to a hospital?
37. Which would you prefer to own, a motorboat or an automobile?
38. Name the process which separates the wheat from the chaff.
39. Is a bee yellow, red, or blue?
40. Why is a steam shovel superior to a hand shovel?
41. For what purpose are lighthouses situated along the shore?
42. Give the names of two patriotic songs.
43. Name an important product of the South.
44. What place of interest would you visit in Wyoming?
45. At what time do most business houses open?
46. Why is the Latin language taught today?
47. Why do we write letters to our friends?
48. Which is heaviest to carry, feathers, iron or coal?
49. Name four presidents of the United States.
50. What office does a push button perform?

INTELLIGIBILITY LIST 47

1. How dense is the vegetation in the jungles of Africa?
2. Of what benefit is paste in a busy office?
3. Of what value is charcoal to a coal fire?
4. What instrument is used to tell time?
5. How does a cold in the head affect the ears?
6. Under what difficulty was the "Star-Spangled Banner" written?
7. How do Indians secure fish in winter?
8. Why is it necessary for women to use hairpins?
9. Name the author of "David Copperfield."
10. Give some ingredients for making fudge.
11. What beverages are drunk throughout the world?
12. Of what advantage is a time table to a traveler?
13. Where do you prefer your vacations, at the seashore or the mountains?
14. Describe the outer shell of the cocoanut.
15. Mention another word for father.
16. When does a man become of age in this country?
17. Give the last letter in the alphabet.
18. Explain how a thumb screw is manipulated.
19. Name some articles that are manufactured from ivory.
20. On what part of the plant do potatoes grow?
21. What pen is best adapted for ruling lines?
22. Name some instruments which compose an orchestra.
23. Characterize the eastern coast of North America.
24. What cruelty did the Indians display in warfare?
25. Why are weeds undesirable in a flower garden?
26. What part does a chorus girl take in a burlesque?
27. What effect did the Erie Canal have on New York City?
28. Of what drink are the English very fond?
29. Name two interesting parks in this city.
30. What state in the Union has the largest trees?
31. What tool is employed by a carpenter to plane a board?
32. What is the advantage of having an ally in warfare?
33. Explain how a mole burrows in the ground.
34. Name some articles manufactured from lumber.
35. Explain why caterpillars are injurious to trees.
36. Do you prefer to live in the city or country?
37. Of what help were the observation balloons during the war?
38. Give the author of the "Village Blacksmith."
39. Describe the appearance of sunshine on water.
40. What kind of instruction is received in a trade school?
41. Describe the color of a harvest moon.
42. Why is there so little plant life in the deserts?
43. Which animal is the larger, an elephant or a lion?
44. What country has six months of darkness?
45. What machine is used for cutting grass?
46. What part of the body is affected by pneumonia?
47. Mention some labor saving machines.
48. Name a city in Italy famous for its palaces.
49. Why should the capital be near the center of the state?
50. Name a plant that is grown from a seed.

INTELLIGIBILITY LIST 48

1. What event is commemorated on the Fourth of July?
2. Where is the Metropolitan Museum of Art located?
3. Name two of our largest warships.
4. Name two of the largest cities in the world.
5. How are freight cars moved across the river?
6. What luck did you have on your last fishing trip?
7. What time of the year are there leaves on the trees?
8. What public parks are located in this city?
9. Name a grain that grows under water.
10. What bell in Philadelphia first rang for liberty?
11. Why are gray squirrels more valuable than red ones?
12. Write down the product of five and fourteen.
13. In what way is the electric fan a benefit in summer?
14. Name a president who has served two terms.
15. Briefly describe Lincoln's boyhood.
16. What is the danger of impurities in drinking water?
17. What thoroughfare is called the "Great White Way"?
18. What invention has increased the speed of traveling?
19. Of what value are awnings in hot weather?
20. In what country are most wild animals found?
21. Why is it hard for horses to draw on wet pavements?
22. From what animals do wool and beef come?
23. Name two ways in which strawberries can be used.
24. Describe the appearance of the skin of a peach.
25. What tool does a carpenter use mostly?
26. Contrast the hoof of a horse and a cow.
27. Why do people paint iron often?
28. Why do people pay fire insurance?
29. For what purpose is a Board of Health established in this city?
30. Was Benjamin Franklin a farmer, statesman, or inventor?
31. Why is the water power from rivers valuable?
32. What newspaper furnishes the name for Times Square?
33. What part of your throat is affected by tonsilitis?
34. What mountains are located in New York State?
35. What dogs played an important part in the war?
36. How is the sewing machine a benefit in the home?
37. Explain why tar paper is used on roofs.
38. Why was "No Man's Land" so desolate?
39. Which is of more use to an engineer, mathematics or Greek?
40. Mention an explosive used in blasting.
41. What birds build their nests in rocky cliffs?
42. What makes an electric fan rotate?
43. What offices are located in the Municipal Building?
44. Write down the names of the five continents.
45. What kind of vessel navigates under water?
46. In what ancient country do bulrushes grow?
47. Name the country in which olives grow.
48. How do we transport food from one place to another?
49. How can you pour a liquid into a bottle without spilling it?
50. Explain the difference between a cafeteria and a restaurant.

INTELLIGIBILITY LIST 49

1. What noted author wrote "Treasure Island"?
2. Why do large stations have an information stand?
3. How does a dentist extract teeth without pain?
4. What fuel is used for oil stoves?
5. Explain why asbestos is utilized to prevent fire.
6. How is an automobile tire filled with air?
7. Explain why office doors are numbered.
8. What name is given to the American Flag?
9. Of what material are expensive umbrellas made?
10. What action made Martin Luther famous?
11. What makes the snow pile up in high drifts?
12. Name the state of which Boston is the capital.
13. Name two fur-bearing animals.
14. Why do firms advertise their merchandise in the papers?
15. Name the author of the "Lady of the Lake."
16. Name a college which deals with military training.
17. Name something which makes you happy.
18. What makes fruit turn red?
19. How can a scholarship be obtained for college?
20. How does turning a ship's wheel steer the ship?
21. What is the advantage of a vacuum cleaner for a carpet?
22. State the number of syllables in an articulation list.
23. Give the abbreviation for the state of Texas.
24. From what country does the best perfume come?
25. What city officials wear brass buttons?
26. Give the advantage of a clip on a fountain pen.
27. Give two uses of the daily calendar.
28. Mention a disadvantage of working in a factory.
29. What crackers are manufactured by the National Biscuit Company?
30. What country completed the work on the Panama Canal?
31. Mention a famous waterfall in New York State.
32. Explain the necessity for filling teeth.
33. What distant objects can be seen through a telescope?
34. What is best for polishing hardwood floors?
35. Which are the larger, field strawberries or cultivated ones?
36. What table-ware first came from China?
37. What use is made of rubber bands?
38. What show has had the longest run on Broadway?
39. Which thermometer is used for scientific work?
40. Which is the equivalent of a shilling in American money?
41. Name a reliable firm that manufactures watches.
42. Why is lead preferred for soldering?
43. Explain the order in which correspondence is filed.
44. What is a convenient method of fastening papers together?
45. What concern manufactures candy at moderate prices?
46. Why is ferryboat traffic difficult in winter?
47. What race of people has kinky hair?
48. The outline of what country resembles a boot?
49. How often does high tide occur?
50. What tears flags on high buildings?

Speech Power and Its Measurement [1]

By L. J. SIVIAN

Bell Telephone Laboratories

The paper is chiefly concerned with the important speech power quantities—frequency spectra, distributions of instantaneous, average, syllabic and peak amplitudes, etc.—as they obtain in actual speech for a large range of voices, talking levels, and subject matters. The analysis is not nearly so complete nor so fine-grained as that which, in principle, can be derived from oscillographic records of individual speech sounds. Its advantage is in the speed with which data can be secured, under widely varying conditions and on a scale which warrants statistical conclusions. Some of the methods in use for measurements of this type are described. A "level analyzer" has been developed, primarily for the measurement of average and peak pressure amplitudes in speech and music, both as to magnitude and as to position in the frequency spectrum. Illustrative results are given for samples of speech, music and noise.

SPEECH sounds are so variable from one to another, from one individual to another, from one conversation to another, as to make it necessary to deal with speech power in a statistical manner. This is particularly true of engineering applications, as distinguished from studies in phonetics and voice dynamics. It is largely from the viewpoint of the former that the subject is here treated.

Speech power occurs in several states, e.g. acoustic, electric, magnetic, mechanical, optical, thermal, etc., but its measurement largely refers to speech in the acoustic or in the electric form. Acoustically, the simplest quantity to define is the instantaneous power transmitted through unit area tangent to the wave front. That power is $L = P \cdot U$, where P is the pressure and U the air particle velocity. With P expressed in bars (dynes per cm.^{2}), and U in cm./sec., L is given in ergs/cm.^{2} $\times$ sec. We do not directly measure the product $P \cdot U$. No suitable means for doing that has been developed. In a progressive plane wave, or with good approximation in any progressive wave at sufficient distance from the source, P and U are in phase, and the expression for L simplifies down to $L = P^2/\rho c = U^2 \cdot \rho c$, where ρc is a constant equal to 41.5 mechanical ohms—the sound radiation resistance of air. Hence for the type of waves mentioned the wattmeter type of measurement (pressure $\times$ velocity, corresponding to voltage $\times$ current in the electrical case) may be replaced with the simpler measurement of pressure (voltage) or velocity (current) alone.

What are the acoustic voltmeters and ammeters that are capable

[1] Presented before Acoustical Society of America, May 11, 1929.

of measuring the instantaneous pressures and velocities respectively? First, as to acoustic ammeters: perhaps the two best known forms are the Rayleigh disk and the hot-wire microphone. The disk gives absolute values of U but its response is so slow that it can be used only to measure the effective values in comparatively sustained sound waves lasting, say one second or longer, or for the ballistic integration of shorter pulses. Hence it does not give a measure of the instantaneous velocities for even the slowest audio-frequency vibration. Furthermore, its use when exposed to the speaker in an open room, is rendered difficult by its susceptibility to spurious air currents. The only application of the disk to vocal power measurements which has come to my notice, is one by Prof. Zernov,[2] published in 1908. For sustained loud singing and shouting he found energy densities ranging from 0.3 to 2.0×10^{-4} ergs/cm.^{3}, at 2 meters distance from the singer. Assuming uniform distribution over a hemisphere of 2 m. radius, this gives a total power output of the voice of the order of 50,000 microwatts. At 2 m. distance reflections from the walls of the room materially raise the energy density as compared with that in a progressive wave.

Another acoustic ammeter is the hot-wire microphone. The resistance variations of the wire tend to follow the instantaneous values of the air particle velocity in the sound wave, but the sensitivity is a complicated function of the frequency, rapidly decreasing as the latter increases. Hence it does not give true oscillograms of speech sounds which in general include a frequency range of six or seven octaves.

Still another device in this general class is the glow-discharge microphone. Its response is determined largely by the amplitude of the air particle motion (E. Meyer, E.N.T., v. 6, 17-21, 1929). Hence, for constant sound intensity the response is inversely proportional to the frequency, roughly. No method for its absolute calibration has been proposed other than comparison with a microphone of known performance. The frequency response and the somewhat erratic behavior of the device in its present status, render it rather unsuitable for speech power studies.

Nearly all our information concerning speech power and the waveform of speech sounds has been obtained with acoustic voltmeters, i.e., with devices responding to pressure in the sound wave. To a first approximation, the ear belongs to this class. It includes the resonators which Helmholtz used in his vowel studies. Generally, the vital element in these devices is a diaphragm which vibrates with

[2] *Ann. der Phys., 26,* 94, 1908.

the alternating sound pressure and whose motion is converted (indirectly, as a rule) into a visual record. These acoustic voltmeters have gradually evolved from Scott's phonautograph and Koenig's manometric capsules and indicating flames of some 70 years ago to the present day technique. Their extensive use accounts for the fact that sound measurements frequently are expressed in terms of pressure rather than in terms of power. In many cases, when the relation between pressure and velocity is not known or too complex a function of frequency to be manageable,—the description of the sound wave must be confined to pressure values alone.

The first requisite for absolute measurements of speech pressures is a microphone which admits of an absolute electroacoustic calibration over the range of audio frequencies, and which has a substantially uniform sensitivity over the most important part of that range. The development of the condenser microphone and of the thermophone method for calibrating it, supplied this need. This fundamental contribution to the subject is due to E. C. Wente.[3] Using such a calibrated condenser microphone in conjunction with a vacuum tube amplifier and an oscillograph which were uniformly sensitive up to about 6000 p.p.s., I. B. Crandall and C. F. Sacia,[4] and I. B. Crandall[4] obtained oscillograms of the fundamental speech sounds in the English language. Essentially, these oscillograms give a picture of the instantaneous pressures throughout the duration of the sound, impressed on the microphone diaphragm at 2.5 cm. from the speaker's lips. A certain amount of similar work on German speech sounds has been published by Trendelenburg.[5] From the standpoint of phonetics these pressure amplitude oscillograms of individual sounds are a most comprehensive source of information. Their chief use has been in determining the frequency spectra of the fundamental speech sounds. Sacia[6] and Sacia and Beck[7] have used them to compute the mean and the peak powers of those sounds.

From the standpoint of engineering applications, speech power has a twofold interest: (*a*), there is the question of designing microphones, receivers, circuits, room acoustics, of controlling noise levels, etc., (*b*), the control of apparatus (chiefly adjustment of amplification) while it is handling speech from persons who acoustically are not controllable.

[3] *Phys. Rev.*, July 1917, April 1922, June 1922.
[4] *Bell Sys. Tech. Jour.*, April 1924 and Oct. 1925, resp.
[5] Wiss. Veroff. a. d. Siemens-Konzern, 1924 and 1925.
[6] *Bell Sys. Tech. Jour.*, Oct. 1925.
[7] *Bell Sys. Tech. Jour.*, July 1926.

In connection with (*a*) we can use data obtained under "laboratory" conditions, i.e. with the speakers, their voice levels, the talking speed, etc., suitably selected. Perhaps the simplest statistical characteristic

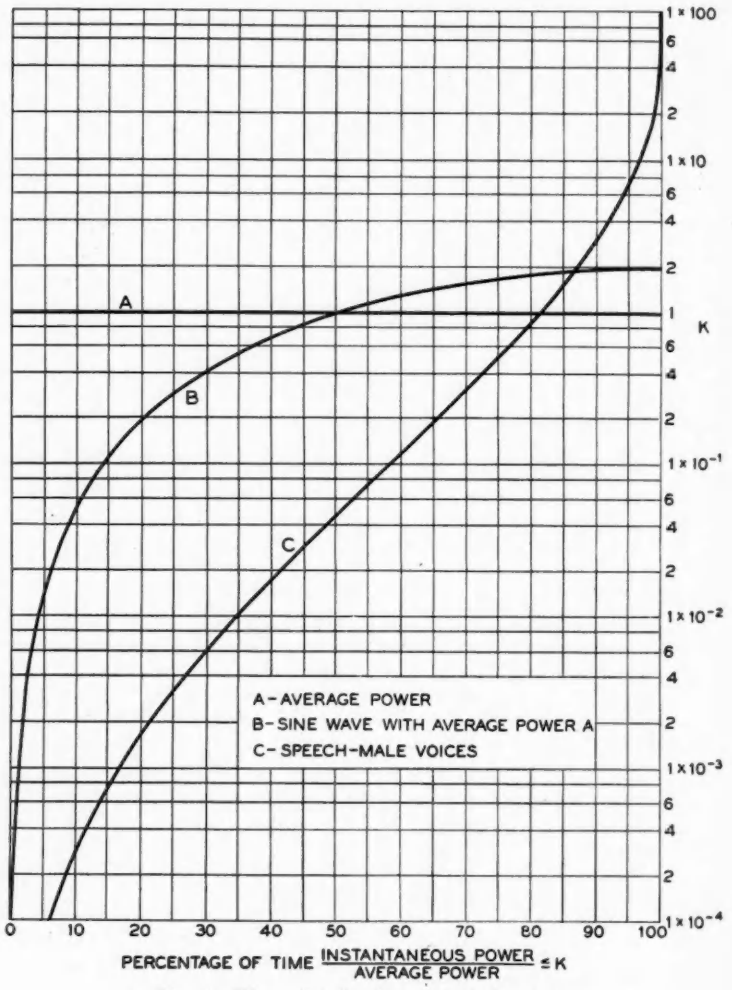


Fig. 1—Time distribution of speech power.

of speech is the average power, i.e. the ratio of the total energy in a large number of speech sounds divided by the total speech time. The quantity immediately measured is the pressure on the transmitter diaphragm. The corresponding power flow through unit area is computed. In doing so it is roughly assumed that the pressure on the diaphragm is twice as great as it would be in free air, which is

assuming total reflection by the transmitter. It is further assumed that the power flow is uniform over a hemisphere whose radius is the distance from the speaker's lips to the diaphragm. For this average power flow with "normally modulated" voices, Crandall and Mackenzie [8] found the value of 12.5 microwatts. More recently Sacia [6] gave 7.4 microwatts, including pauses between speech sounds, which

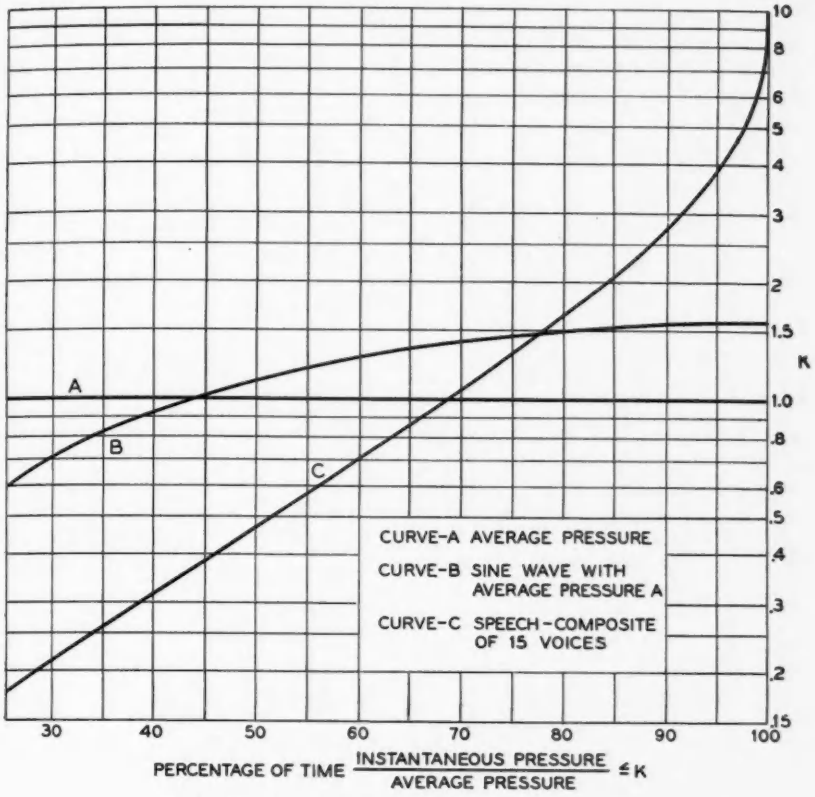


Fig. 2—Time distribution of speech pressures.

occupied about one-third of the total time. Some measurements made since then have led to substantially similar values, and so the value of 10 microwatts may be taken as fairly representative of a normal level. The corresponding average pressure on the diaphragm, at 5 cm. distance from the lips, was found to be about 5 bars. The averaging, of course, is for the absolute values of the pressures. Some notion of the level involved may be gained by noting that it is

[8] *Phys. Rev.*, March 1922.

decidedly fatiguing to maintain a speech level 20 db higher for more than a few seconds.

It should be noted that the distributions of the instantaneous power and instantaneous pressure values in respect to their average values are quite different for speech from what they are in a sinusoidal wave. Fig. 1 and Fig. 2 show the extent of the difference for powers and pressures respectively. Clearly, speech as a whole is decidedly more "peaked" than a sinusoidal wave. Values of peak factors for

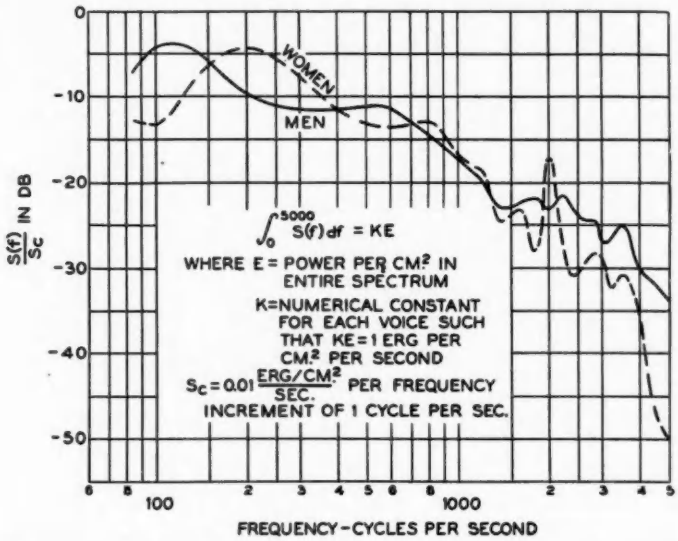

0
-10
-20
-30
-40
-50
S(f)/Sc IN DB
WOMEN
MEN
∫₀⁵⁰⁰⁰ S(f)df = KE
WHERE E = POWER PER CM² IN ENTIRE SPECTRUM
K=NUMERICAL CONSTANT FOR EACH VOICE SUCH THAT KE=1 ERG PER CM² PER SECOND
Sc = 0.01 ERG/CM²/SEC. PER FREQUENCY INCREMENT OF 1 CYCLE PER SEC.
6 8 100 2 3 4 5 6 8 1000 2 3 4 5
FREQUENCY-CYCLES PER SECOND

Fig. 3—Energy spectrum of connected speech-composite curves for 4 men and for 2 women.

individual speech sounds, i.e. the ratio of peak amplitude to mean effective amplitude, are given in Sacia's paper.[5]

A paramount characteristic of speech is the distribution of power and pressure in the frequency spectrum. That is so because in general the systems responding to or carrying speech (including the ear) have pronounced frequency characteristics. The same is true of noise sources interfering with speech. Here, as for total powers and pressures, the simplest quantity is the average spectrum for a large number of speech sounds. Crandall and Mackenzie[8] obtained the energy spectrum shown in Fig. 3, based on speech from six voices. They used a condenser microphone whose output was analyzed by a series of resonant circuits covering the range from 75 to 5000 p.p.s. Fig. 4 shows a recent determination in which the average pressure is given as a function of frequency. The apparatus

used to obtain the data will be described in detail elsewhere. Briefly, a series of band-pass filters are used to separate the frequency components. The output of any one filter goes into a rectilinear vacuum

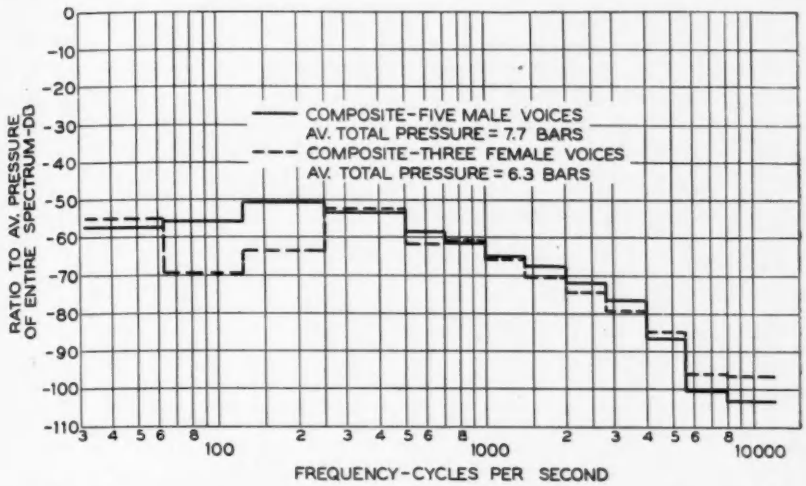


Fig. 4—Average speech pressures per frequency interval of 1 cycle per second—normal conversational voice. Distance 2″.

tube rectifier, and a fluxmeter integrates the rectified current over the duration of the speech. Simultaneously with this measurement the total spectrum also is rectified and integrated. The integration periods used were 15 seconds long. The pressure in any one band is

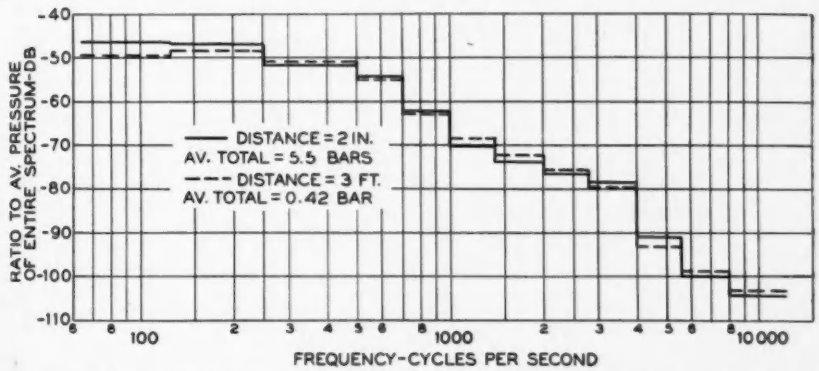


Fig. 5—Effect of distance on voice spectrum—average pressure per frequency interval of 1 cycle per second—normal speech-composite, three male voices.

given by the product: ($10^{0.05\alpha}$ × band width in cycles × total pressure), where α is the ordinate expressed in decibels (db). In using average spectra of this sort it is well to remember that they are

determined not only by the amplitudes in the individual speech sounds but equally by frequency of occurrence. Thus the fact that nearly all vowels for male voices at normal levels have a fundamental frequency between 80 and 150 p.p.s. tends to accentuate that region even though the fundamental amplitudes in individual speech sounds not be outstandingly large.

The data in Fig. 4 are for close talking conditions, 5 cm. from the lips to the diaphragm. Fig. 5 shows the effect of distance on the spectral distribution. The two distances are 5 cm. and 90 cm. respectively, in both cases the transmitter being set into a large fiber-board wall. The shapes of the two spectra are almost identical

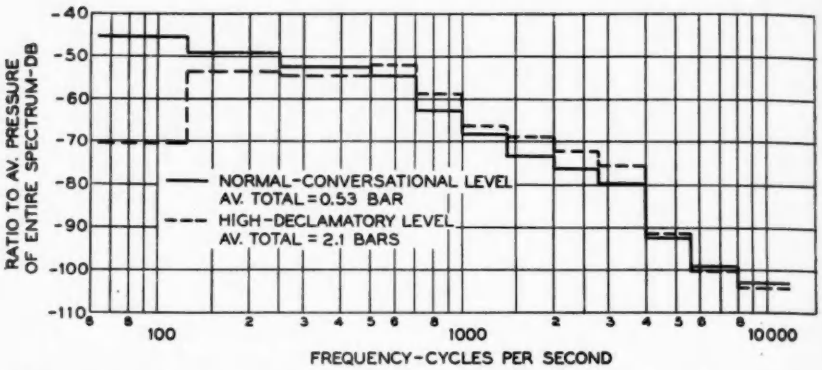


Fig. 6—Effect of level on voice spectrum—average pressures per frequency interval of 1 cycle per second—composite, three male voices, distance 3 ft.

indicating that, on the average, even at 5 cm. the condenser microphone does not greatly affect the voice as a sound generator. The largest difference between the two is in the lowest band, from 62 to 125 p.p.s., perhaps owing to the relatively low radiation efficiency of the voice at those frequencies.

The ratio of the distances is 18 : 1, that of the average pressures 14 : 1. The average pressure is nearly inversely proportional to the distance, part of the difference probably being chargeable to the more nearly total reflection for the distant condition. It is implied, of course, that even for the latter condition the direct sound is large compared with that reaching the microphone by reflections.

So far the spectra discussed were those of normally modulated voices. Fig. 6 shows what happens to the average spectrum when a high, rather declamatory level is adopted. The higher level is relatively poorer in frequencies below 500 p.p.s., relatively richer between 500 and 4000 p.p.s., and above 4000 p.p.s. its spectrum is nearly the

same as for normal levels. The decrease at low frequencies is most pronounced in the band from 62 to 125 p.p.s., i.e. in the region of the voice fundamental frequency at normal levels. This leaves two possibilities. Either at the high speech level the fundamental frequency is the same as for normal, but its amplitude is relatively small; and the spacing of the overtones is the same as at normal levels. Or else—and more probably as indicated by the auditory pitch sense —passage to the high level is attended by an actual upward shifting of the fundamental frequency, and a correspondingly larger spacing

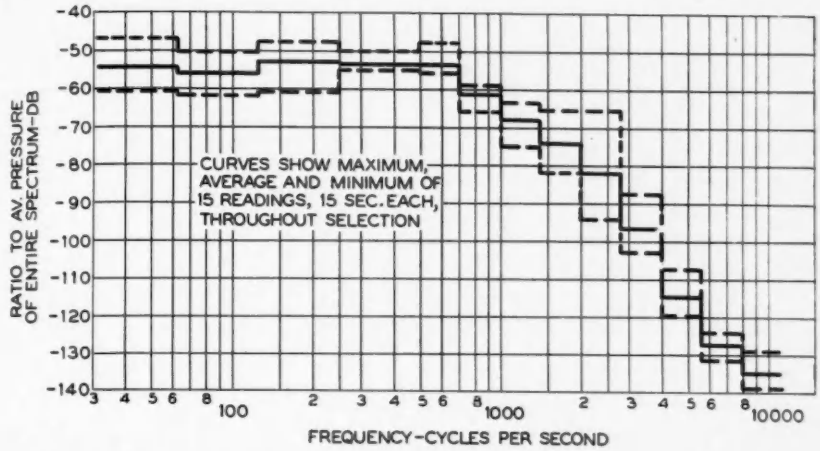


Fig. 7—Average pressures per frequency interval of 1 cycle per second—piano selection—Liszt's "Hungarian Rhapsody No. 2"—average total pressure = 3.5 bars.

of the overtones. In addition the amplitude of this new fundamental would be relatively lower than for the normal level. The average spectrum method is incapable of deciding between the two alternatives. Some oscillograms were made at normal and high speech levels, which clearly indicated that the second alternative is the correct one, or at least the prevalent one. For most vowel sounds, the fundamental frequency was found shifted from about 100 p.p.s. at normal to about 200 p.p.s. at the higher level. This result has an intimate bearing on the loss of naturalness encountered when speech originally picked up at normal voice levels is subsequently reproduced at much higher levels. A corresponding change, though in the opposite direction, probably takes place in going from normal to subnormal levels. Some evidence of this will be seen in the section on peak amplitudes.

The method of average pressure-frequency spectra is equally

applicable to sounds other than speech, provided they are sustained or can be repeated. Two illustrations are given. Fig. 7 is for a piano composition. Characteristic in comparison with speech is the relative smallness of high frequencies. Spectra of radically different types of compositions were found to be rather similar. The instrument, the room acoustics and the position of the microphone largely determine the picture. In this case, the microphone was about 7 ft. from the keyboard, and the reverberation time between 1.2 seconds at 100 p.p.s. and 1.0 second at 4000 p.p.s.

Fig. 8 gives the spectrum of street noise entering a laboratory

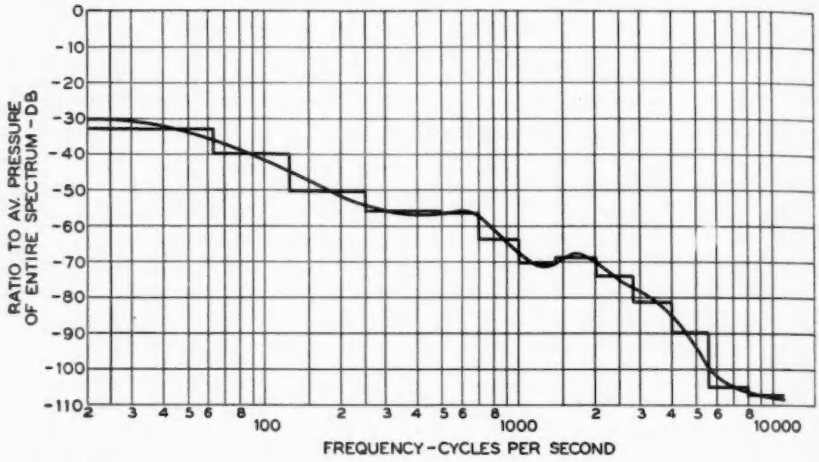


Fig. 8—Average pressures per frequency interval of 1 cycle per second—street noise—10th floor Bell Telephone Laboratories facing east—2.50–3.20 P.M., Oct. 10, 1928—average total pressure = 0.75 bar.

window, ten floors above the street. Street traffic and an elevated railway are the chief contributors, the measurements being taken during traffic peak hours. The relative poverty in high frequencies is even more pronounced than for the piano.

We shall now consider the type of apparatus intended primarily for measurements when the speakers are not acoustically controllable. Instead of averaging over a large number of words the measurement is essentially that of mean power in syllables. Usually it covers the whole spectrum rather than a particular frequency band. It has been widely used for controlling amplification in radio broadcasting, in phonograph and film recording of speech, and for rapid measurement and electrical control of speech levels in telephone conversations. A typical device of this sort is the "volume indicator" [9] shown in

[9] Essential features of this apparatus are shown by E. L. Nelson, U. S. Patent No. 1523827, filed 8/31/22.

Fig. 9, developed about ten years ago. Essentially it is a vacuum tube rectifier with a rapid action d.c. meter in the plate circuit. It is operated on a part of the characteristic such that the rectified plate current is roughly proportional to the square of the speech voltage. The rectifier is preceded by an amplifier of adjustable gain. For the speech level under measurement the gain is adjusted to such a value that the fluctuating meter deflections attain a prescribed

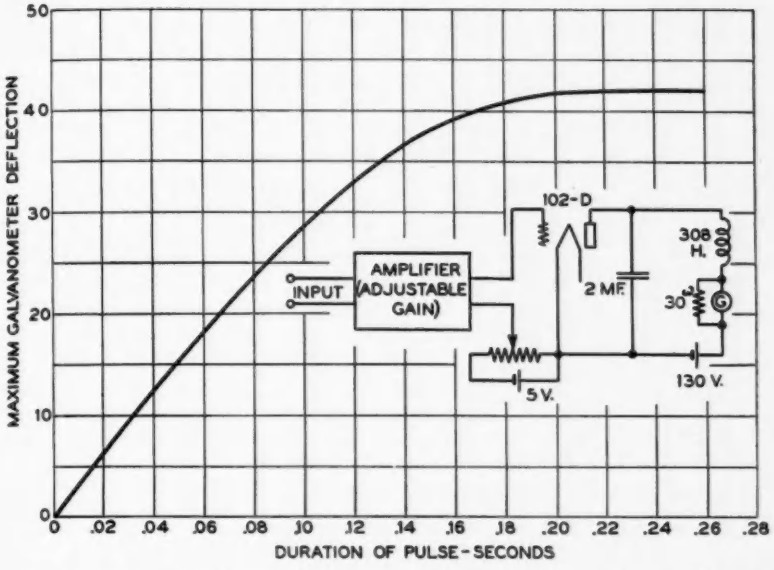


Fig. 9—Volume indicator—deflection as a function of a.c. pulse duration.

maximum value on the average of once in about three seconds. That value of gain, expressed in decibels with respect to a certain normal value of the gain, gives the volume indicator measure of the speech level. The meter, combined with the electric circuit, has a dynamic characteristic as shown in the curve, which gives the maximum deflection as a function of the duration of the a.-c. input. For inputs lasting more than about 0.18 second the maximum deflection remains the same. Since the average syllable duration is of the order of 0.2 second, it follows that the maximum deflection of the "volume indicator" meter is approximately proportioned to the mean power in the syllable. By comparison with oscillograms, or by equivalent methods, the volume indicator readings can be correlated by absolute quantities. Fig. 10 is an example showing the relation between sepech level as measured with the volume indicator and the average

instantaneous amplitude of the speech waves on a certain laboratory telephone circuit of the commercial type.

Fig. 11 illustrates the type of data which can be expeditiously secured by means of the volume indicator. Here the levels for a

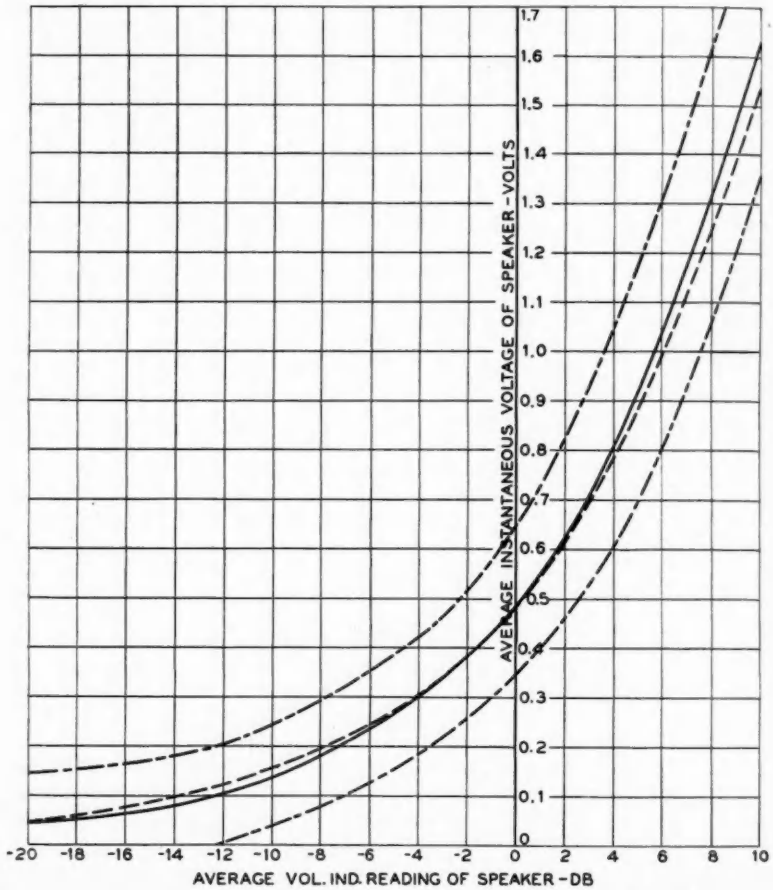


Fig. 10—Instantaneous voltage V.S. Volume indicator reading 85% of observed points included between outer dashed curves—ordinates of middle dashed curve are inversely proportional to gain of volume indicator.

large number of speakers and conversations (over a laboratory telephone circuit of the commercial type) are determined. To cover the same ground by means of oscillographic measurements would have been a decidedly formidable task.

Another device for the rapid measurement of speech levels is the "impulse meter," [10] shown in Fig. 12. This is essentially a peak

[10] D. Thierbach, *Zs. F. Techn. Physik*, No. 11, 1928.

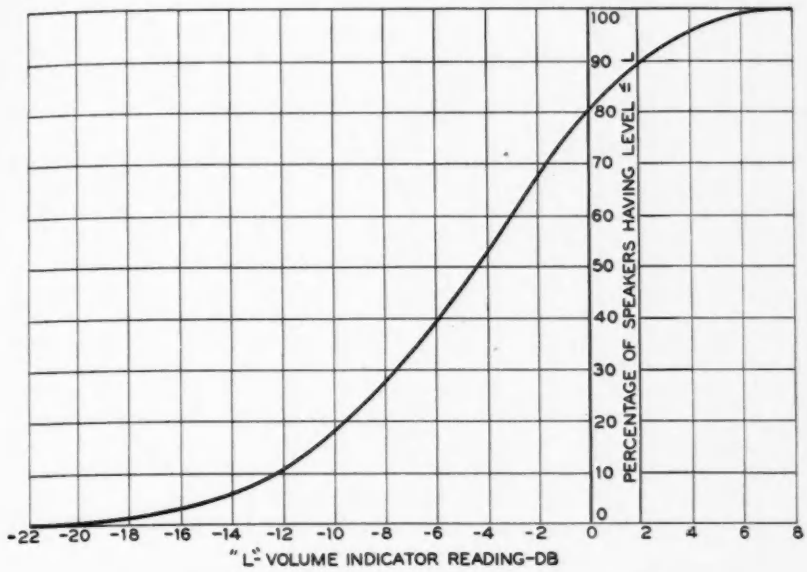


Fig. 11—Distribution of speaker's levels—87 men and 59 women.

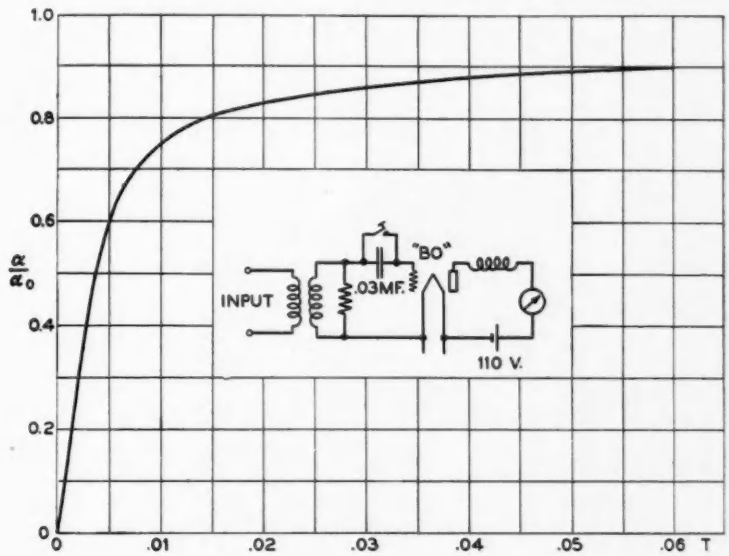


Fig. 12—Impulse meter.

T = Duration of a.c. impulse
α = Corresponding ultimate plate current
α_0 = Ultimate plate current for steadily applied a.c.

voltmeter. The curve shows the time rate at which the potential on the blocking condenser builds up. The time required for the galvanometer to reach its maximum deflection is determined by the dynamic characteristic of the meter and associated plate circuit, as well as by the time constant of the condenser charging circuit.

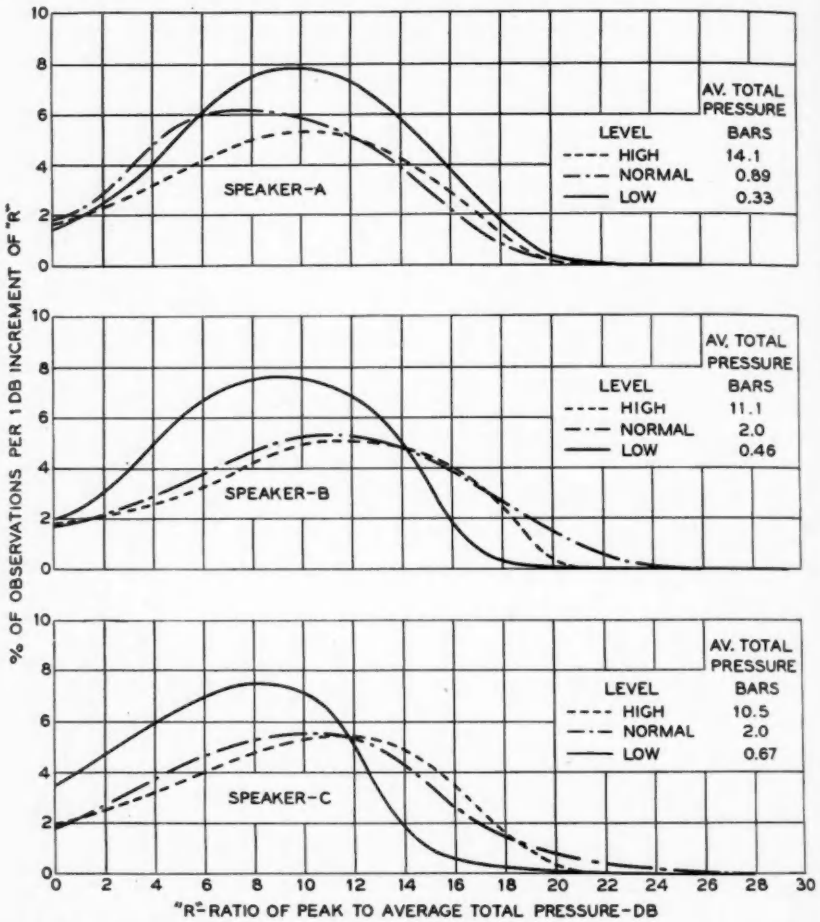


Fig. 13—Distribution of peak pressures for entire speech spectrum—three male voices—each peak is the maximum instantaneous pressure during 1/8 sec.

In using the volume indicator type of instrument in the telephone plant, limits are observed which have been determined by trial to give satisfactory operating results. The fact that it is possible to set such limits indicates a correlation between mean syllabic power and peak pressures. The overload capacity of apparatus is so important that

laboratory investigations of quantities affecting it merit more fundamental study than can be made with instruments of the volume indicator type. The quantities of interest are the instantaneous amplitudes of the wave peaks, the frequency of their occurrence and their distribution in the frequency spectrum. The apparatus used to obtain the data described below will be described in detail elsewhere. It is sufficient to state here that it is capable of directly (and automatically) registering the magnitudes and frequency of occurrence of the in-

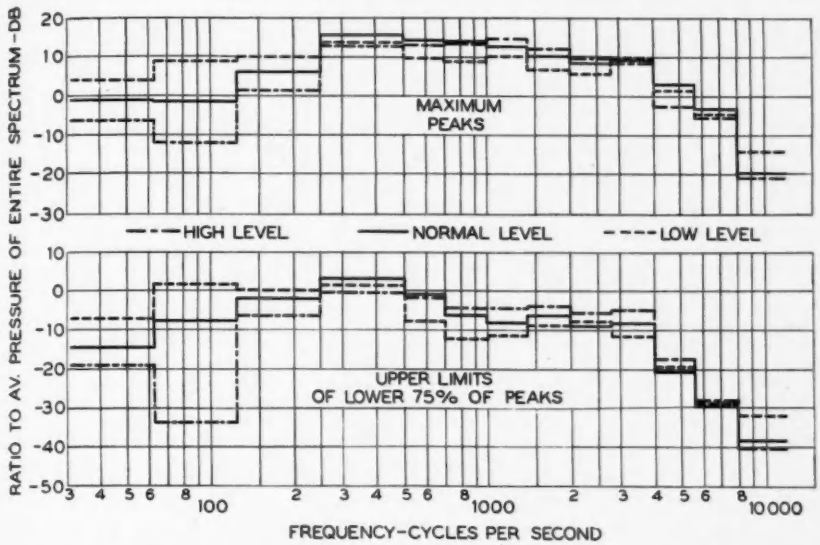


Fig. 14—Peak pressures of speech—composite, three male voices—each peak is the maximum instantaneous pressure during 1/8 sec.—average total pressure:
High (declamatory) level—8.7 bars;
Normal (conversational) level—1.6 bars;
Low (confidential) level—0.5 bar.

stantaneous peaks over a range of 60 db, corresponding to a power ratio of 1,000,000 : 1. The peak amplitudes were measured for the whole spectrum and also for restricted frequency bands selected by filters.

Fig. 13 and Fig. 14 show the results of some measurements with undistorted speech. Each individual observation gives the magnitude of the peak pressure in a 1/8 second interval. This is about as short an interval as one can use and still retain a high degree of probability that the individual measurements will give the maximum peak amplitudes in syllables. Otherwise, from the standpoint of the apparatus, the individual observations could be confined to much shorter time intervals, resulting in many more measurements for a

given length of speech. The ordinates give the ratios of the peak pressures to the average total speech pressure. In Fig. 13 the peaks are those of speech as a whole; in Fig. 14 the ordinates give the peak pressures in the several frequency bands. This is done for three widely different average levels spread over a range of 30 db. From

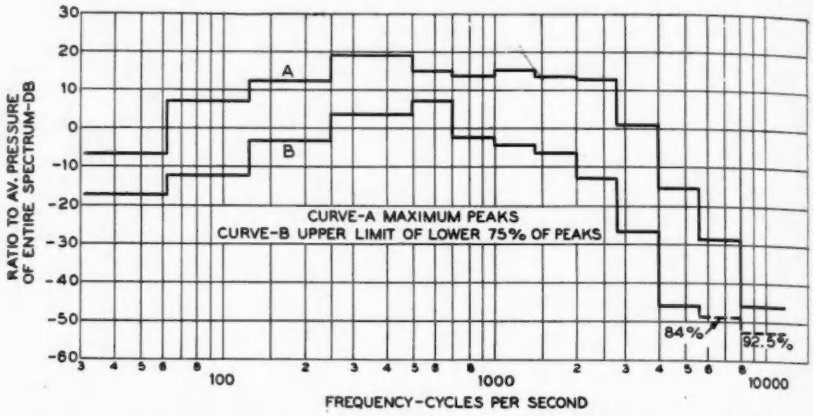


Fig. 15—Peak pressures in piano music—Liszt's "Hungarian Rhapsody No. 2" —each peak is the maximum instantaneous pressure during 1/8 sec.—average total pressure 4.0 bars.

the standpoint of providing apparatus overload capacity it is sometimes important to know not only the maximum values of all peaks (which might be uneconomical to provide for) but the upper limit of a certain percentage of all the peaks. The lower half of Fig. 14 illustrates this method of analyzing the peak data. It is interesting to note how much larger the transmission capacity must be to take care of the highest 25 per cent of the peaks.

Fig. 15 gives a similar picture of the peak amplitude distribution in a piano composition, for which the average amplitudes are given in Fig. 7. The microphone was about 7 ft. laterally from the center of the keyboard, and the measurements were made in a room having an average reverberation of about 1 second.

My thanks are due to Dr. H. K. Dunn and Mr. S. D. White of Bell Telephone Laboratories, who have obtained most of the data discussed in this paper.

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THE THEORY OF SOUND PERCEPTION

By G. E. Shambaugh, M.D.
Chicago

The earliest theory of sound preception of which we have record is the theory of "aer implantus" propounded by Aristotle in the fourth century, B.C. This theory was based on the assumption that since sound results from an air-born impulse, it required a cavity in the head filled with air to respond to these impulses originating in the outer air. This air, implanted by the Creator, was the actual end-organ of hearing. It was not until 1760 that Cotugno discovered that the labyrinth of the ear is filled with fluid, not with air. Long before this, however, the physiological importance of the air supposed to be implanted in the labyrinth had been discarded. Various ideas were advanced regarding structures supposed to exist in the internal ear which should respond in vibration to sound impulses. By some it was believed that the terminal filaments of the eighth nerve were these vibrating structures. Others held that "sonorous bands," not nerve filaments, stetched across the cavity of the cochlea were set into vibration. The idea that there exists in the cochlea a vibrating mechanism which responds to impulses of tone waves was clearly set forth in the writings of Claude Perrault in 1680. Duverney in 1683 was the first to liken the cochlea to a musical instrument which responds in different parts to tones of different pitch according to the principles of physical resonance. He believed that the lamina spiralis was the vibrating mechanism and because this bony plate is broader in the basal coil and becomes gradually narrower towards the apex of the cochlea, he located the

perception for the low tones in the basal coil and for the higher tones in the apex.

Scarpa, in 1798, discovered the existence of the membranous labyrinth, and Corti, in 1851, discovered the details of the end organ in the cochlea, the Organ of Corti. With the discovery of these details came the conviction that the anatomical structure in which the physical impulses of sound waves were transformed into nerve impulses, is the hair bearing cells which receive the terminal filaments of the acoustic nerve. The problem of sound perception became simplified into determining the manner in which these hair cells are activated. Helmholtz, the great physicist, accepted as the method of bringing about the stimulation of the hair cells, the principle of physical resonance, which had formed the basis of theories of sound perception for at least 200 years before his time. He approached the problem from the stand-point of the physicist, elaborating the resonance theory to explain the various phenomena of sound perception, and then sought to find in the cochlea a structure which the physicist could accept as a resonating mechanism. His first conclusion was that the rods of Corti, varying in size as do all the structures in the cochlea, from the base to the apex, filled the role of a vibrating mechanism, and in vibrating jar the hair cells which are in proximity and bring about their stimulation. When he was shown that crocodiles and birds, which possess excellent hearing, do not have these rods, he turned to the idea of string resonators and selected the basilar membrane as the vibrating structure. The radiating fibres of this membrane stretch across the cochlear tube from the lamina spiralis to the spiral ligament and are longest at the apex, becoming gradually shorter toward the base. The low tones he believed set to vibrating the radiating fibres of the basilar membrane at the apex and the high tones, at the base of the cochlea.

Since the time of Helmholtz various theories have been advanced to explain the phenomena of sound perceptions, each theory starts with hypothesis that the basilar membrane is the vibrating structure. Ewald, for example, did not accept the idea of physical resonance. He substituted the theory that each tone in the scale produced undulations in the basilar membrane throughout the entire length of the cochlea. These undulations would be different for each tone of the scale and since it was believed that the membrana tectoria was separated by an appreciable space from the projecting hairs of the hair cells, only the hair cells which occupied the crests of these undulations would brush against

the tectorial membrane and be stimulated. A series of groups of hair cells from one end of the cochlea to the other, would thus be stimulated by each tone. And since this group complex would be different for each tone it would form a suitable basis for tone perception and peripheral tone analysis. In 1907 I demonstrated that the membrana tectoria is not separated from the hairs of the hair cells but is adherent to them. This fact renders the Ewald theory untenable for although different tones in the scale might produce each its own particular type of undulations in the basilar membrane, with the hairs attached to the membrana tectoria, every tone would produce undulations throughout the entire length of the cochlea and thus stimulate every hair cell of Corti's organ. This would leave no basis for peripheral tone analysis fundamental to any theory of sound perception.

Careful study of the structure of the basilar membrane makes one doubt if this membrane is capable of filling the role of the vibrating mechanism which brings out the stimulation of selected groups of hair cells for each tone in the scale. The first requirement of such a vibrating structure is that it should respond in the same way at all times to the same impulse, otherwise we would have no ability to recognize pitch, and diplacousis would be a constant phenomena. Now the fact that the basilar membrane is a vascular structure and that these vessels must dilate and contract as do all blood-vessels, makes it impossible for the membrane to vibrate the same way at all times to the same impulse. It is hardly satisfying to assert that because the membrane is suspended in fluid, its vibrating qualities are not altered by being altered by being weighted differently at different times. There is another structural reason for rejecting the basilar membrane as a vibrating structure. As one examines this membrane toward the proximal end of the basal coil, it is found to lose all resemblance to a vibrating structure and becomes a thick triangular wedge in shape, and this too, just where it should become thinner and shorter to fill the role of a string resonator.

There are two methods of approach to the problem of sound perception. One is that of the physicist who examines the phenomena of hearing, elaborates a theory that will best explain these phenomena and then searches the anatomy of the cochlea to discover a structure which will satisfy the requirements of this theory. This was clearly the method pursued by Helmholtz. It is my feeling that in accepting the basilar membrane as a vibrating structure, we may have been following a false lead.

The other approach to this problem is the biological approach. By this method we make a study of the structures in the cochlea, endeavor to ascertain what may be the probable function of the different structures and then proceed to elaborate a theory of hearing that will fit in with these conclusions. Of these two methods of approach, the latter would seem to be more reasonable.

The first fact that impresses the anatomist in studying the minute structures of the labyrinth, is that while there are three types of end organs in the internal ear, the Organ of Corti in the cochlea, the macula acustica in the utricle and saccule and the crista ampullaris in the semicircular canals, they are structurally fundamentally the same. In each of these end organs the structure of paramount physiological importance is the hair bearing cell. In each there is superimposed above the hair cells, an epithelial structure to which the hairs are attached. In the organ of Corti, this is the tectorial membrane, in the macula it is the otolith membrane, and in the crista it is the cupula. In each one of these end organs the stimulation of the hair cells is dependent upon an interaction between their projecting hairs and this superimposed structure. In two of these end organs, the macula and the crista, this interaction can be brought about only through the movements of the superimposed structure. It would seem logical therefore, to conclude that in the remaining end organ, the one found in the cochlea, the stimulation of the hair cells of Corti's organ is also dependent on movements of its superimposed structure, the membrana tectoria, movements which are transmitted to this membrane by sound impulses passing through the endolymph. It would seem logical, therefore, that speculations regarding the stimulation of the hair cells of Corti's organ should begin with this biological conclusion, that the membrana tectoria is the vibrating mehanism, the movements of which activate the hair cells. To set aside the tectorial membrane as some do, because it does not appear to satisfy the requirements of a physical resonator, is not satisfactory. In the first place, I would point out that the membrana tectoria is an elusive structure to study and as yet we do not know enough concerning its physical properties to warrant the conclusion that it may not act as a resonator. In the second place, I would suggest that should substantial evidence be forthcoming that the membrana tectoria cannot act as a physical resonator, then the logical procedure would be to discard the resonator hypotheses and work out some other method of response to explain the various phenomena of sound perception.

I might be permitted here to offer some speculations regarding the possible way in which the membrana tectoria might respond to sound impulses. In the first place, I would point out that the membrana tectoria, like the other structures in Corti's organ, varies in size from one end of the cochlea to the other, being considerably larger toward the apex of the cochlea and diminishing to a much tinier structure near the proximal end of the basal coil. The membrane is securely attached to the vestibular lip of the sulcus spiralis internus along the outer margin of the lamina spiralis. It is also anchored to the supporting cells just mesial to the inner row of hair cells by the structure recognized as the streifen of Hensen. It is not a homogenous gelatinous structure as it is sometimes supposed. Sections cut directly across the coils of the cochlea demonstrate striations which radiate out from the mesial attachment of this membrane to its outer lower border. These striations are not in the nature of fibrillae as is readily apparent in sections made through the membrane when it is folded. At no place does one ever find the cut ends of filaments. The striations are formed by a sort of lamellar structure. From the variations in size of this membrane it would seem that impulses from the tones highest in the scale passing through the endolymph might suffice to throw into vibration only that part of the membrane which is very tiny, found at the proximal end of the basal coil, and that each tone lower in the scale would be able to activate a larger and larger area of the membrane, until the lowest tone which we can hear would throw the entire membrane into undulations. Such a theory it would seem might account for the recognition of pitch and a peripheral tone analysis. A fundamental defect in this hypothesis is the results obtained by experimental studies in which circumscribed areas in Corti's organ are made to undergo degeneration through over-stimulation by a particular tone, while Corti's organ remains unimpaired both above and below this area. A second method of response which one might postulate is that the entire membrane is thrown into undulations by each tone of the scale, the undulations differing of course for each particular tone. There are palpable objections to this hypothesis as well, since we know that the hairs are attached to the membrana tectoria, every tone would have to produce stimulation of all the hair cells, which leaves no basis for peripheral tone analysis and no explanation for areas of degeneration in the midst of Corti's organ produced by over-stimulation of certain tones. The third method of response that occurs to one is that areas in different parts of the cochlea are stimulated by each tone. This hypothesis comes

nearer to accounting for the known facts regarding sound perception both in normal hearing and in defects in hearing. It accounts also for the production of areas of degeneration by over-stimulation of particular tones. It is apparent from the examination of the extent of these areas of degeneration that the vibration of the membrane in response to a particular tone extends over a considerable area, but all that is required to account for the phenomena of tone perception and peripheral tone analysis is that a somewhat different grouping of hair cells be stimulated by each tone of the scale.

In conclusion, I would point out that the proper approach to the problem of the physiology of hearing is the biological approach; that when we study the problem from this point of view the membrana tectoria appears to be unquestionable the logical structure for taking up the impulses that pass through the endolymph and for transferring them to the hair cells. The manner in which it performs this role is the fundamental problem. One should be ready to accept that manner of response which best explains the phenomena of hearing.

PSYCHOPHYSIOLOGICAL CONSIDERATIONS RELATING TO THE THEORY OF HEARING

By Leonard T. Troland
Technicolor Motion Picture Corporation

I. Introductory

Theories of hearing are largely attempts to explain the correlations which hold between aspects of *auditory experience* and the characteristics of the *acoustical stimuli* which impinge upon the ears. From a modern point of view, the auditory experiences are *psychological* factors, whereas the stimuli are *physical*. In endeavoring to present the psychological facts of audition, analytically, we distinguish between a number of supposedly distinct *attributes* of auditory experience, and we hope to ascertain the stimulus features upon which they depend. It is natural that psychological and physiological investigators should emphasize the subjective factors, while more physically minded students tend to start with variables which are directly demonstrable in the stimulus. Thus, from the physical standpoint, one recognizes, primitively, the relationship between pitch and frequency, and that between loudness and amplitude at any single frequency. Noise quality and space localization also receive attention, but such features of auditory experience as volume and brightness, together with fusion and over-all quality, have been left largely to the psychophysiologists. However, a comprehensive theory of hearing must obviously deal with all of the variables, both subjective and objective, in the situation.

Classical psychophysiology seems to be correct in its view that auditory—like all other forms of—experience is correlated directly only with neural activities occurring in that portion of the brain which is known as the cerebral cortex. The domain of the acoustical stimulus, on the other hand, may be regarded as terminating at the ear-drums, where it first acts upon the physiological mechanisms. From the tympanum to the cortex, however, is a rather "far cry" (if we may stick to acoustic language), and the theory of hearing seems to be an effort to describe the apparatus and functions which intervene between these two points. The theory naturally involves linking the stimulus characteristics, on the one hand, with the sense-organ processes of the ears, and correlating the attributes of auditory experience, on the other hand, with features of auditory nerve excitation. In establishing these

linkages, we must have recourse to hypotheses, but our speculations must be consistent with the data of anatomy and of physiology.

It is not my purpose in the present paper to attempt a review and criticism of the many theories of hearing which have been advocated, from the time of Helmholtz down to the present time. This would indeed be too large a task, so I shall confine myself to a much smaller and decidedly less significant intention, that of reviewing certain theoretical possibilities which I have already discussed elsewhere, but which seem to have been inadequately emphasized hitherto. I ought to say, before beginning this review, that the ideas which I shall consider are very similar to, although somewhat more in detail than, those suggested in the same connection by Dr. Fletcher in his recent book. I had not seen his discussion when I was first led to some of these conceptions.

I shall consider primarily the relationships which may hold between the qualities and attributes of auditory experience and the functions of the afferent nervous arc. In doing this, I shall assume that the mechanics of the sense-organ response are essentially the same as those described by Fletcher. These constitute what may be characterized as a modern interpretation of the Helmholtz resonance theory, and they provide explanations for certain very important features of auditory response, including: (1) the audibility curve, (2) tonal analysis, (3) subjective harmonics, difference and summation tones, and (4) masking. However, these ideas concerning the mechanisms of the middle and inner ear do not provide us with a perfectly clear picture of the physiological functions which underlie even such fundamental auditory attributes as pitch and loudness. Their implications regarding noisiness, volume, brightness, consonance and dissonance are even less definite. It is to a further expansion of the neural and subjective relations of this theory that I shall devote my attention.

II. The Psychoneurology of Pitch and Loudness

Let us consider, first, the psychoneurology of pitch and loudness, regarded either as attributes of tones or of noises, but primarily of the former. Psychophysiological hypotheses regarding pitch have been classified into "frequency theories" and "place theories," according as we make pitch a function of the location of the response along the cochlear canal or of the oscillatory frequency of the response process. Doctrines of the type advocated by Helmholtz and by Fletcher have

usually been associated with the "place theory," because the stimulus frequency actually determines the position of the response. However, I have found certain advantages, which I shall endeavor to indicate, in combining the resonance mechanism with the view that, after all, subjective pitch is determined by the *nerve impulse frequency*, rather than, primarily, by the correlated "position."

As an approach to this view, we may consider the fact that auditory space localization, on the basis of phase differences between the stimuli at the two ears, seems logically to necessitate an actual representation of the stimulus time pattern in the auditory nerve currents. We know, from general neurology, that these currents must be pulsatory in character, and there seems to be no way in which to represent phase differences except by differences in the actual timing of the pulses in the two auditory nerves. This conception has been considered in detail by Banister, who assumes that the nerve impulses are set off at the hair cells at a fixed phase in the cycle of vibration of the basilary membrane. It follows that, at least up to a frequency of about 500 cycles (corresponding to the lowest limit set by the refractory phase constants of nervous tissue) the stimulus frequency is actually reduplicated in the auditory nerve currents and, hence, in all probability, in the auditory brain process. Consequently, up to the limiting frequency in question, it would seem legitimate to make pitch depend psychophysically upon the frequency of the nerve process in the auditory section of the cerebral cortex.

If we reject this conclusion, we must inquire what contribution impulse-frequency actually does make to the auditory consciousness. It would be in harmony with current neurological principles to suppose that frequency is a component of the intensity feature of the response, and hence that it is contributory to *loudness*. However, loudness must be variable as a function of stimulus intensity independently of frequency, and, to make this possible, varying numbers of coöperating fibres must be called into play, since the pulse frequency in each fibre is fixed by the frequency of the stimulus. This means that loudness depends not upon the frequency *per se*, but upon the gross number of pulses delivered per second by the entire system of fibers which is operative at the given frequency. I believe that this is the most acceptable hypothesis regarding loudness; and it obviously does not prevent us from regarding the abstract frequency characteristic of the process as being the determinant of pitch.

Now, the idea that pitch is directly correlated with nerve pulse frequency encounters a number of rather forcible objections. Perhaps the most important of these lies in the apparent limitation on nerve pulse rate which is imposed by "refractory phase." The lowest duration of "absolute refractory phase" which I have yet heard quoted authoritatively would not permit the transmission, along a single auditory fiber, of an impulse frequency greater than 2000; while figures ordinarily quoted indicate a limit of about 700. Personally, I consider it entirely possible that the auditory fibers have very special properties in this respect, and it is a fact that we know nothing directly concerning the refractory characteristics of these or any other human sensory nerves. However, there is another way around the difficulty, which seems to me to be extremely interesting and to have escaped the attention of auditory theorists.

It is generally achnowledged that resonance, within the inner ear, cannot be limited to the response of single basilary and nerve fibers, even in the case of pure harmonic stimuli. At all points above the absolute intensity threshold—and, perhaps, even at the threshold—a *group* of approximately synchronized nerve currents must be involved. Now, it should be obvious that the frequency limitation set by refractory phase applies only to *single* fibers, and not to a group of fibers acting in coöperation. In such a group, a rhythmic alternation of the transmission between the members can carry a much higher frequency. This effect is actually demonstrated, in a striking manner, by electroscope records made by Adrian on the sensory nerves of the frog. If we assume, as we must, that the relation of pitch to frequency involves the integral properties of the nerve current arriving at the cortex along a bundle of fibers, there seems to be no objection to supposing that frequencies greatly in excess of those set by the refractory limit of a single nerve fiber can be transmitted from the sense-organ to the cortex. I am, therefore, inclined to reject the refractory phase argument as irrelevant until it can be made more detailed and quantitative.

Another class of objections to a combination of the resonance theory with a frequency hypothesis for pitch alleges that the hypothesis in question is redundant, in the face of the actual pitch analysis which takes place upon the basilary membranes. However, a careful study of the facts shows that the redundancy is only apparent, not actual. I must say, in the first place, that subjective analysis—as signalized by Ohm's law for audition—should certainly be attributed to the separation of harmonic and other components along the cochlear canal and

not to a sorting of neural frequencies in the cortex. It does not follow that, because the cortex actually receives some of these frequencies, it has any mechanism for separating them. Its capabilities will probably be taxed to the utmost to *keep* them separate. Harmonic analysis is one thing and pitch discrimination is quite another. Now, Wegel and Lane have shown that pitch discrimination is so fine that it corresponds to a displacement of the resonance maximum along the basilary membrane of approximately 0.02 millimeters, or to a space sensibility about one hundred times greater than that of touch at the finger-tips. Knudsen's data indicate that the brain can detect a displacement corresponding to only nine auditory fibers, while earlier measurements indicate even smaller values. These facts must be combined with others which show that the resonance peaks upon the basilary membrane are extremely blunt, at least in the lower frequency range. This bluntness is demonstrated by the quantitative facts of masking and by the necessarily high damping of the inner ear resonators. The facts of masking show that, even with a 1200 cycle note, there is appreciable interference at basilary positions corresponding to 600 and 3000 cycles, a span of more than one-third the total length for the system.

It is evident that any "place theory" of pitch must provide some means whereby the position of maximal response, within such wide bands, can be determined with great delicacy by the brain. At first thought, such a mechanism seems inconceivable, because of the "all or none" law for nerve conduction. This necessitates that all of the co-excited fibers shall have the same response amplitude, so that the brain receives the excitation as a plateau pattern rather than as a peak. As we shall see later, there may be some features of the nerve current which tend to indicate the position of the maximum excitation but, on the whole, it would seem to be imposing a very difficult task upon the cortex to base the pitch impression upon a triangulation of such incidentals. Why should it not directly utilize the actual stimulus frequency, which is most certainly represented in the total neurological disturbance?

The objector may still feel inclined to question the utility of resonance in the sense-organ, if pitch and pitch discrimination can be determined in the simple manner above indicated. In further reply to such objections, I should like to emphasize the following point, which seems hitherto to have escaped adequate notice. ***Resonance is powerfully instrumental to increase in sensitivity, as well as to analytical separation.*** If the auditory sense-organ were entirely non-resonant, its

sensitivity could certainly not be as great as we find it to be at its best. Upon a sensitivity basis alone, evolution would surely have produced an even more resonant system, if a reasonably powerful damping were not required in the interests of acoustic definition in the temporal dimension. However, the partial sorting out of different frequencies along the length of the basilary membrane, and hence among different groups of fibers within the auditory nerve, permits a transmission to the brain of a representation which is more adequate than would otherwise be possible. The characteristics of complex sounds could not be represented with fidelity if they all had to be integrated within the temporal responses of a single nerve fiber excitation or those of a synchronized group of fibers.

I have dealt more in detail with some of these problems in a previous paper,* where I have indicated certain experimental observations which might enable us to test the correctness of these interpretations. It is, of course, not necessary to assert that "place" has no influence whatsoever upon pitch judgment or discrimination. Since "place" is, at least crudely, a function of frequency, we should expect the brain mechanism to take it into consideration along with frequency. Nowever, its influence would be rough, as in judgments of "absolute pitch", rather than delicate, as in experiments upon pitch discrimination where different stmuli are presented in close succession. The only phenomena of which I am aware that are difficult to reconcile with the above views are those of double disharmonic hearing, in which the two ears respond to a single harmonic stimulus so as to yield two different pitches. On a frequency theory, this phenomenon would evidently require a departure of the nerve pulse frequency from that of the stimulus for at least one of the ears.

III. The Psychoneurology of Volume and Brightness

I shall now consider briefly the manner in which the views above advocated enable us to deal with the characteristics of auditory experience which are known as volume and brightness, respectively. These attributes of noises, as well as of pure tones, have received relatively little attention from physicists, but have been studied in detail by certain psychologists. Volume is recognized as a function of stimulus frequency, being large for low-pitched sounds, and diminishing with

* Troland, L. T. The Psychoneurology of Auditory Qualities and Attributes. Journ. of Gen. Psychol., 1929, vol. 2, pp. 28–58.

advancing pitch. It also increases with loudness. It is definitely a spatial attribute of the auditory experience. Brightness, on the other hand, stands for the shrillness or penetrating quality of the experience, and tends to increase as the pitch becomes higher. Ogden has suggested that brightness corresponds to a certain "saliency" or pointedness of the auditory quality. It would be my view, however, that this pointedness exists in the dimension of harmonic analysis, rather than in that of auditory space.

It is in harmony with certain suggestions which have been made by Boring that we should correlate volume with the spatial size or extent of the cortical region which is involved in the given auditory excitation. It is natural to suppose that such cortical areas will be larger, the greater the number of coöperating afferent fibers. Now, if different pitches are presented with equal loudnesses, the number of fibers must be greater for low than for high pitches, since—in accordance with our previous assumptions—equal loudness must involve the same gross number of impulses per second for each of the frequencies which is represented. We thus reach a conclusion which is in harmony with the recognized increase of volume with decreasing frequency.

Brightness can be regarded as being associated with the sharpness of the resonance peaks upon the basilary membrane, such sharpness increasing rapidly with increasing frequency. It is, of course, necessary to have some representation of this saliency in the afferent nerve currents. Such a representation can perhaps be found if we suppose that even when all of the fibers in the given nerve group are responding to their maximal extent (according to the "all or none" principle) and approximately in synchronism, there may nevertheless be some minor differences between the nerve excitations corresponding to different portions of the excited band. For example, we should expect the impulses which are released at the resonant center of the band to be somewhat in advance of those which are released from the edges of the band. This effect would generate a sort of curvature of the neural wave-front, and the degree of such curvature would furnish a possible neurological basis for the attribute of brightness. This hypothesis appears to be in harmony with the variations of brightness among complex sounds, such as vowels, as well as a function of pitch.

IV. Correlations of Non-Tonal Phenomena

The general view of auditory psychophysiology above advocated furnishes a very satisfactory basis for dealing with phenomena of non-

tonal auditory quality, including noises and vocables. In general, we may attribute tonality to the rythmicity or degree of rhythmicity in the neural discharges in a group of associated auditory conductors. Noisiness, on the other hand, may be correlated with the degree of departure of the nerve currents from such temporal uniformity and synchronism. The over-all form quality (*Gestaltqualität*) of any auditory experience would rest upon an integration of the temporal and spatial patterns of the composite excitation which arrives at the brain areas. The simplest deviation from pure tonality yields the attribute of *timbre*, which recognizedly depends upon the distribution of overtone intensities along the basilary membrane. Pure timbre, without noisiness, may be supposed to involve the excitation of a series of non-overlapping bands or groups of fibers. Overlapping may, however, be permitted if the frequencies are harmonically related, so as not to interfere or beat with one another.

When inharmonically related frequencies overlap, the beating action disturbs the rhythmicity of the neural response within the overlap band, and the resulting irregularity introduces a noise factor into the corresponding experience. When the interfering frequencies are only slightly different, as in the ordinary experimental demonstration of "beats", the irregularity is only slight, and appears, primarily, as an intermediate frequency, corresponding to the familiar subjective "intermediate tone." The latter, naturally, "carries the beats". With greater frequency separation, the irregularity must become more marked, as will appear from a study of the complex wave-forms which are produced by such combinations. It is, of course, evident that the pulsatory nerve currents cannot reduplicate such wave-forms in their totalities, but will only pick out their more salient features, in a manner depending upon the absolute intensity of the stimulus. (Incidentally, we can conceive of no mechanism by which the cortex could reconstruct the original vibration from its fragmentary representation within the nerve current; and, *a fortiori*, no way in which it could analyze out the components of the original stimulus.)

It is well-known that the stimuli to noise are lacking, either partially or wholly, in harmonic composition, and hence may be expected to yield radically overlapping excitations upon the basilary membrane, followed by irregularly timed nerve currents in the auditory nerve fibers. However, no matter how complex the stimulus may be, its various frequency components will be segregated to some extent along the cochlear structure. Some noises may be confined exclusively to low-

frequency, others exclusively to high-frequency regions, and there may even be unexcited bands between low- and high-frequency components of certain noises. However, within the zones of excitation, there will be a failure of the basilary and naural responses to conform to a strictly rhythmic pattern, because the spread of the theoretical components is too great, in relation to their proximities, to permit complete analytical separation. It is in harmony with these considerations that noises should possess a modified attribute of pitch, corresponding to average frequency values or ranges, but also that they should have further attributes—depending upon the degree and character of the non-rhythmicity. Since departures from regularity of timing can be of an almost infinite number of different kinds, we should expect to find the wide variety of noise qualities which actually appear in experience. The loudness of a noise will, of course, be associated with the total number of auditory nerve impulses arriving at the cortex from all coexcited receptors, while volume and brightness attributes will likewise be determined in the same manner as in the case of pure tones.

Vocables, including vowels and consonants, rest upon specific nerve excitation patterns of the sort above suggested. The vowels are, in general, more tonal than are the consonants, and they possess specific pitch attributes corresponding with the frequency distributions of their stimuli. The consonants are relatively more noisy, but also lie in characteristic pitch regions.

V. Conclusion

In concluding, I may summarize the view which I have presented, as follows:

The general picture of the inner ear mechanism which is advocated by Dr. Fletcher and his associates has been adopted as a starting point for the purpose of tracing the probable relationships between experiential auditory attributes and the characteristics of the afferent nerve currents. It is supposed that, although a certain degree of harmonic analysis in the sense-organ actually correlates certain stimulus frequencies with the excitations of distinct groups of nerve fibers, pitch is nevertheless determined by the nerve impulse frequencies within these fibers, rather than by their anatomical identities. The refractory phase characteristics of a single nerve fiber do not limit the transmittable frequency, because different members of a fiber group can act in alter-

nation and still maintain a definite integral rhythm. The attribute of loudness is associated with the gross number of impulses reaching the brain in unit time along all of the coöperating fibers. Auditory volume, on the other hand, is determined by the anatomical number of the fibers which are coöperatively excited. Auditory brightness is referred to the sharpness of curvature of the impulse wave-front within the fiber group. Noisiness, or lack of tonality, is attributed to the existence of a non-rhythmicity or irregularity in timing in the nerve impulses, this being due to an overlapping and interference of different frequencies upon the basilary membrane. The over-all auditory quality is correlated with the total temporal and spatial pattern of the currents arriving at the cortex within any short period of time.

A SPACE-TIME PATTERN THEORY OF HEARING

By Harvey Fletcher
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In constructing any theory of hearing, six classes of quantitative information must be considered, namely, (1) the limits of the pitch and the intensity that the hearing mechanism is capable of handling, (2) the minimum changes in pitch and intensity that are perceptible, (3) subjective tones, (4) the masking effects of one sound upon another, (5) the relation of loudness to sensation level, and (6) binaural effects. Besides these effects there are other auditory phenomena which cannot be expressed in as quantitative a manner but they are just as important. The principal ones are (1) the ability of the ear to analyze a complex tone into its component tones, (2) the effect upon the pitch, the loudness, and the quality of a musical tone when certain groups of its components are eliminated, (3) the relation of brightness and volume (as used by psychologists) to frequency and intensity of the stimulating tone, and (4) the effect of shifting the phases of the components of a musical tone. It is assumed that the reader is familiar with these various effects. They will be referred to at the appropriate time as the theory of hearing is developed.

Two general types of hearing theories have been put forth from time to time to explain these effects. One might be called a space pattern theory and the other a time pattern theory. In the first theory, it is assumed that the time pattern of the wave motion in the air is transferred into a space pattern in the inner ear so that the nerve impulses reaching the brain give us information concerning the time pattern of the wave motion by means of the location of the nerves which are stimulated. In the second theory, it is assumed that the time sequences are transmitted directly to the brain. It is the opinion of the author that both of these effects are operating in aiding one to interpret the sounds which one hears. The term "A Space-Time Pattern Theory of Hearing" therefore best expresses this conception.

When a sound wave is created in the air, the air particles oscillate back and forth and cause small pressure changes which occur in a certain time sequence depending upon the kind of sound created.

If a small[1] pressure gauge which follows instantly the pressure varia-

[1] This size must not be too small or the Brownian Movements would be superimposed upon the wave motion.

tions of the air, were placed at a certain point in the path of the wave, its fluctuations for the three classes of sound indicated, would be as shown in Fig. 1. Each curve is a pressure time pattern of the sound it represents. This pattern is transmitted through the air and communicated to the drum of the ear. It is then transferred through

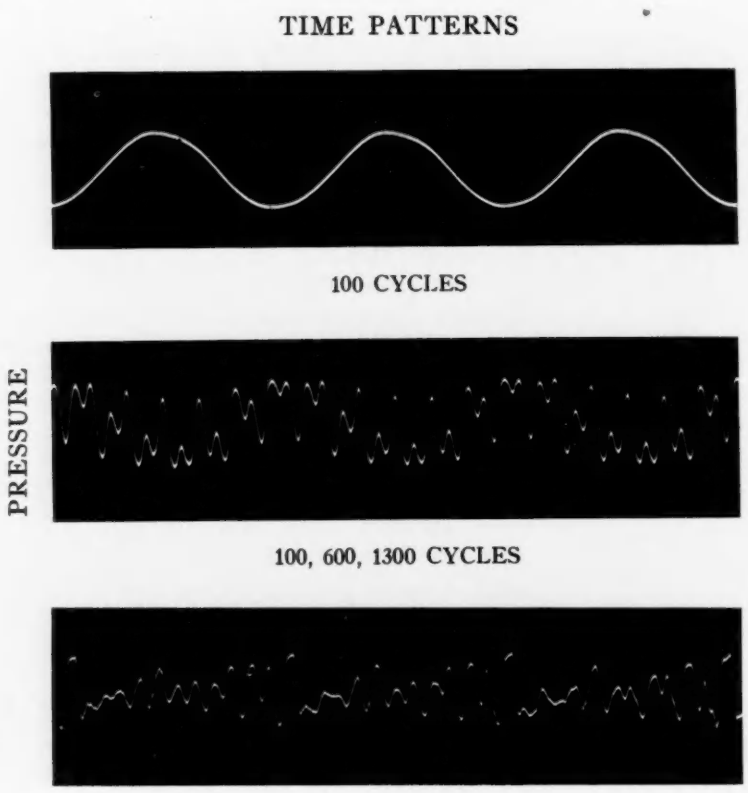


FIG. 1

the ossicles of the middle ear to the liquid of the cochlea. In this process of transmission, the pressure changes are usually considered to be magnified about 60 times. This is due to the lever action of the ossicles and also because the total force[2] exerted upon the ear drum must be distributed over the much smaller area of the oval window. If we knew

[2] This is only true provided that the ear drum does not break up into segments as it vibrates.

all of the mechanical constants involved we could predict the other changes in the pattern. For low intensities and for sounds containing components having vibration frequencies in the middle range, the time pattern probably undergoes only slight modifications. Components having frequencies above or below this range are probably transmitted less efficiently. In any case there will exist a definite time-pressure pattern in the liquid of the cochlea near the oval window which is similar to but not exactly of the same form as that existing in the air.

Non-Linear Characteristics of the Middle Ear

If the intensity of the sound is increased without otherwise changing the pattern in the air, there is a change in the pattern transmitted to the cochlea not only in magnitude but also in form. For high sound intensities the elastic members in the middle ear will exceed the amplitude limits within which the strain is proportional to the stress. Also, the limit for extension is different from that ~~than~~ for compression. Under such circumstances it is well known that frequencies other than those in the original pattern are introduced. For example, for such high intensities a pure tone of frequency f will be transmitted to the cochlea as a tone having components of frequency f, 2f, 3f, 4f, etc. These components which are added to the original tone in the process of transmission account for the subjective tones. Their magnitude can be measured experimentally as follows. Introduce the pure tone into the ear at a known intensity level. At the same time introduce another external tone which differs in frequency from the harmonic being measured by two or three cycles. Beats will then be produced between the exploring tone and the subjective harmonic. If the intensity of the exploring tone is adjusted until the most distinct beats are obtained, then its intensity is a measure of the magnitude of the subjective harmonic. In Fig. 2 are shown the magnitudes of such harmonics for various intensities of the stimulating sound. These curves were constructed from some experimental measurements made by Mr. F. H. Graham. The abscissae give the number of the harmonic[3] and the ordinates the intensity level. For example, a tone having a fundamental[3] at zero level (1 microwatt per square centimeter) will produce subjective harmonics having levels of -1.4, -2.6, -3.7, and -4.8 bels for the second, third, fourth, and fifth harmonics, respectively. It is difficult to determine the exact intensity of the subjective tones, yet within the

[3] The fundamental is called the first harmonic.

observational error and individual variation, it is found that this set of curves fits the experimental facts for tones of all frequencies. This is what one might expect, since the overloading of the middle ear depends upon the physical intensity of the impressed sound rather than upon its sensation level.

Further it might be expected that this overloading would be dependent upon the amplitude of vibration of the ear drum and the chain

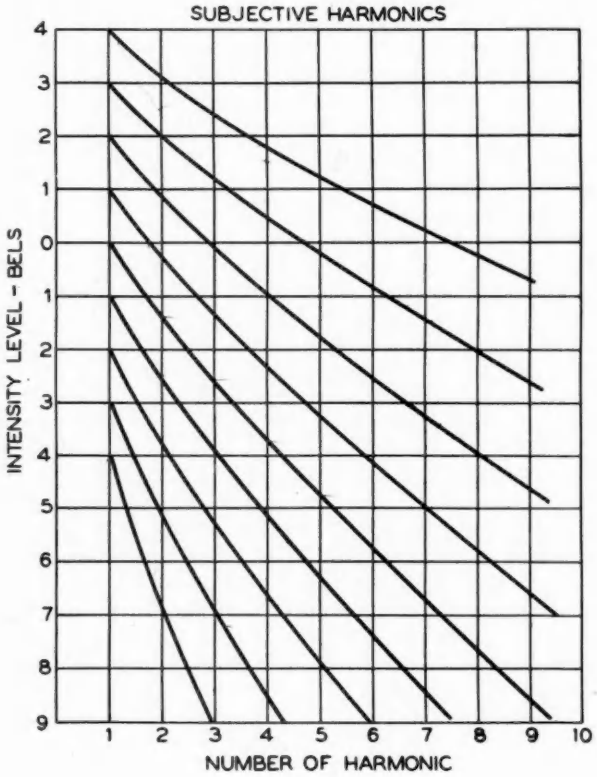


FIG. 2

of bones attached to it. To be in agreement with the above experimental facts it is evident that the impedance at the ear drum must be largely in the nature of a stiffness rather than a resistance or a mass. Recent data by Tröger[4] show that the impedance of the ear is of this character except for high frequencies. It is important to remember that the curves in Fig. 2 are for intensity levels rather than for sensa-

[4] The Reception of Sound by the Outer Ear, Phys. Zeit., Vol. 31, no. 1, Jan. 1930, p. 26–47.

tion levels. Expressed in terms of sensation levels the magnitudes of the subjective harmonics for the tone given above, when its pitch is −4 octaves, would be 4.4, 4.6, 4.4, 3.8, and 3.0 bels above the threshold. It is seen that the first harmonic has a higher sensation level than the fundamental. For still higher intensity levels several of the harmonics may have higher sensation levels than the fundamental.

In Fig. 3 are shown the observational points for four different levels of a tone having a frequency of 50 cycles per second. As you will notice the first six overtones for the higher levels have a sensation level higher than the fundamental. It must be remembered that the data are limited and tests upon any one ear give results which differ appreciably

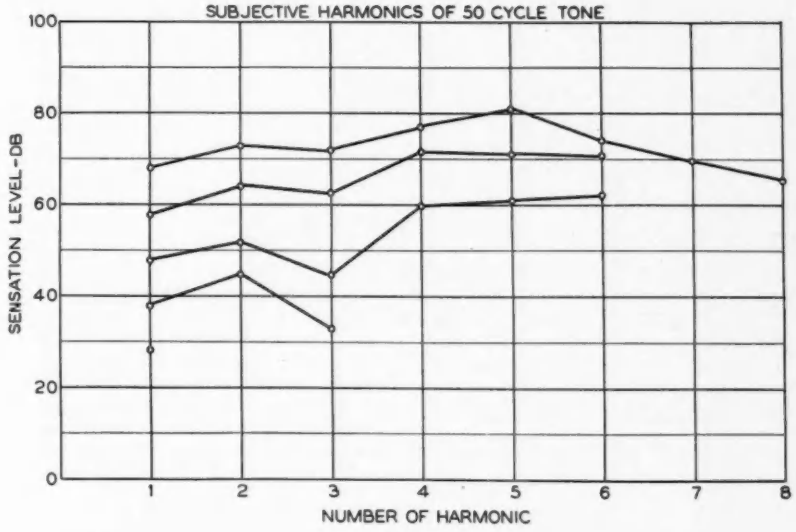


FIG. 3

from average curves. Due to this non-linearity of the middle ear transmitting mechanism the ratio of the amplitudes in the time-pressure pattern in the cochlea to the corresponding ones in the air in front of the ear drum will be much less for the high than for the low intensities. As the intensities increase the transmitting system becomes more and more overloaded and consequently the output diminishes comparatively.

For intensity levels below −6 bels, the output is proportional to the input. For higher levels the output drops below the one to one ratio. If during the vibration, the displacement at the oval window is the same on either side of the neutral position, then only the odd harmonics

would be present. Since both even and odd harmonics are found the vibration must be unsymmetrical and a rectifying action must take place. Thus we have both dissymmetry and non-linearity in the process of transmission.

MECHANICS OF THE MIDDLE EAR

To simplify the kind of motion taking place in the cochlea, we shall first consider the case when no subjective harmonics are present. Let a pure tone be transmitted through the middle ear to the fluid in front of the oval window in the cochlea. The question then is what will be the motion of the fluid in the inner ear and in particular what will be the motion of the basilar membrane where the nerve endings are located.

At the outset let me emphasize that no satisfactory solution of this problem has been made nor is a solution offered here. Wegel and Lane[5] in their paper on "The Auditory Masking of One Pure Tone by Another and Its Probable Relation to the Dynamics of the Inner Ear", made a good approach to the problem when they suggested an electrical analogue of the cochlea. However, they made no numerical calculations due to the uncertainty of the constants involved. In spite of the many objections which may be raised to the simple treatment given below still it probably helps one to better understand the action of the cochlea.

According to the Helmholtz resonance theory, the vibration communicated to the liquid from the stapes is transmitted a certain distance through the scala vestibuli depending upon the pitch of the tone, and then through the basilar membrane, back through the scala tympani to the round window. The nerve endings at the position where the vibration is transmitted through the basilar membrane are stimulated and send impulses to the auditory nerve which carries them to the brain.

Consider a small element of the basilar membrane of length δx and width w and located at a distance x from the oval window. The sinusoidal pressure applied at the oval window is transmitted through the liquid and causes this element to vibrate. The problem then is to determine the magnitude of its vibration.

If the variation in pressure is slow all surfaces of equal area in the cochlea will experience the same force at practically the same time. Consequently, the forces on either side of the element under consideration are equal and opposite and would therefore keep the element

[5] Phys. Review, Feb. 1924.

stationary. As the pressure variation becomes more rapid, there will develop an excess pressure on the upper side which will cause the element to move. The force F due to this excess pressure will be

$$F = k_F w \cdot \delta x P \cos \omega t \tag{1}$$

where P is the pressure amplitude at the oval window and $\omega/2\pi$ is the frequency of variation.

For the reasons given above we have introduced the parameter k_F which varies from unity at high frequencies to zero at very low ones.

The reactive forces opposed to the impressed force mentioned above are due mainly to the inertia and frictional resistance of the liquid moved and the stiffness of the basilar membrane at the point considered.

The effective mass is due almost entirely to the liquid on either side of the element. It seems reasonable to suppose that this effective mass m is given by

$$m = k_m w \cdot \delta x \cdot x. \tag{2}$$

Stated in words this means that the effective moving mass of liquid is proportional to the distance the element is away from the oval window and proportional to the amount of liquid in contact with the element. It is evident that this would be true for the simple type of motion depicted in Fig. 4, where a column of liquid reading from the oval window to the position x and back on the other side of the

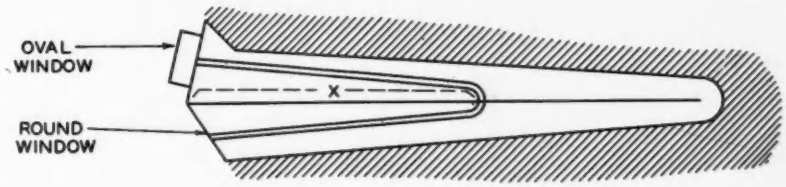


FIG. 4

membrane to the round window, and having a cross section $w \cdot \delta x$, moves to and fro as a unit. In this case $k_m = 2$ since the density of the liquid is approximately unity. Although the motion cannot be as simple as that described yet the general patterns of the motion due to tones of different frequencies are probably somewhat similar and differ only in size. For these reasons one might expect Equation (2) to be approximately correct.

The stiffness S is principally due to the element itself, although of course there is some stiffness offered to the movement of the liquid

by the round window and the oval window and the bones attached to it. The stiffness factor is proportional to the number of cross fibres in the element δx, to their average tension T, and inversely proportional to their length w, that is the width of the membrane at the position x. Therefore

$$S = k_s \frac{\delta x T}{w} \cdot \tag{3}$$

It is well known that such a system has the natural frequency f_0 given by

$$\omega^2 = (2\pi f_0)^2 = \frac{S}{m} = \frac{k_s}{k_m} \frac{T}{w^2 x} \cdot \tag{4}$$

This then is an equation which gives a relation between the position x on the basilar membrane for maximum stimulation and the frequency f_0 of the tone impressed upon the ear.

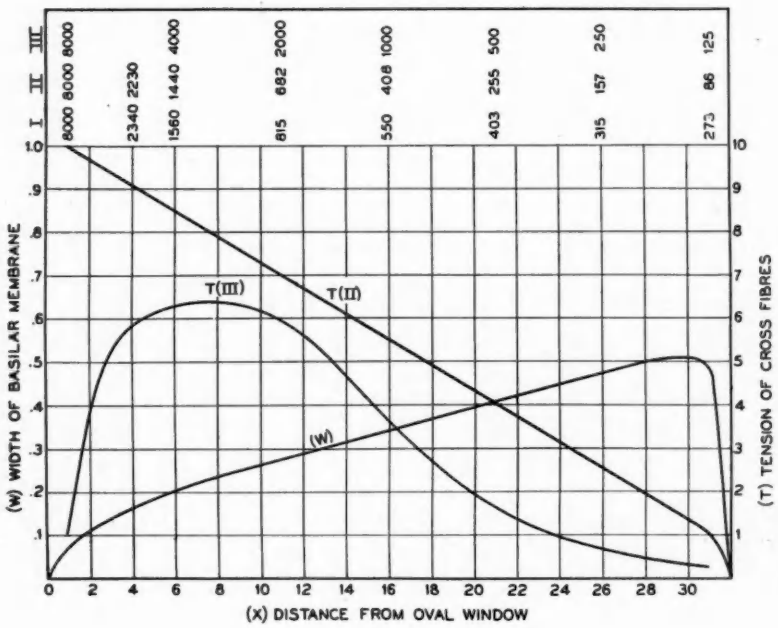


FIG. 5

Let us first consider the anatomical data in connection with equation (4) to see what they would predict concerning the positions of maximum stimulation for tones of different pitch.

The width w can be obtained fairly accurately from microscopic

cross-sections. Although there is considerable variation from sample to sample, the curve marked "w" in Fig. 5, may be taken as representing an average result. It is seen that w varies from .08 to .5 as x varies from 1 to 31 mm.

The anatomical data indicate that the tension T of the cross fibres decreases as x increases but no quantitative data are available. Three cases are postulated and the calculated results are shown in Fig. 5. In Case I, it is assumed that the tension of the cross fibre is constant throughout the length of the membrane. In Case II it is asssumed that the tension decreases uniformly with the ratio of 10 to 1 in going from the oval window to the heliocotrema. In Case III the vibration of T is chosen so as to give a uniform distribution along the basilar membrane of the pitch covering six octaves. In all cases k_s/k_m was determined by assuming that for the resonator at $x=1$ the resonant frequency was 8000 cycles per second.

It is seen from the anatomy of the cochlea that we would expect the position of the maximum stimulation for the high frequency to be near the oval window and move toward the helicotrema as the frequency is lowered. It is seen that the necessary range of resonance may be covered without making any radical assumptions regarding the variation of the tension of the cross fibres. Even though no variation in the tension occurs still the range of resonance is from 273 to 8000 cycles per second.

Most authors who have tried to make a simple analysis of this problem have assumed that the frequency of resonance is inversely proportional to the length w of the cross fibres. but this analysis makes it inversely proportional to w^2. This is because it was assumed that the effective mass due to the vibrating liquid which is associated with a small portion of the basilar membrane, decreases as w decreases. In other words, the effective vibrating volumn is just sufficiently wide to cover the basilar membrane. A ten-fold variation in T accounts for a variation in f_0 from 86 to 8000 cycles per second. If we assume that the rods of Corti extend to values of $x=.05$ cm. and that w varies as indicated, then this range will be extended to 16000 cycles per second. This is only true provided that the equation can be applied to this end condition, an assumption which is probably not true.

The experimental data on minimum perceptible differences in pitch[6] indicate that the distribution frequency along the basilar membrane

[6] This was pointed out by Wegel & Lane in the paper already cited.

corresponds more nearly to that of Case III. For this reason this simple relation between f_0 and x will be used in the following discussions. Mathematically stated then, the resonant pitch

$$P_0 = \frac{14}{5} - \frac{x}{5} \tag{5}$$

or

$$\log_{10} f_0 = \frac{16 - x}{16.6} + 3 \tag{6}$$

which holds for values of x from 1 to 31. According to this view then, the positions of maximum response are distributed uniformly along the basilar membrane according to the pitch of the exciting tone and cover a range of six octaves from -3 to $+3$. Impressed tones having pitches above or below this range do not create a condition of resonance in the cochlea. This may be one of the reasons why the sensitivity drops off for tones having pitches beyond these limits.

Having established the relation between f_0 and x, let us now consider the form of vibration of the basilar membrane. If we were dealing with separate stable resonators instead of mutually connected unstable ones, the problem would be simple. But as we have seen this is not the case. The stiffness of a given portion of the membrane remains essentially constant for impressed tones of different frequency but the effective mass vibrating with it and also the resistance r changes.

The subsidence of the vibration after the force stops is dependent upon the ratio of the resistance-r to the effective mass m, that is, upon the damping factor which is given in bels per second by

$$\Delta = .434 \frac{r}{m} \cdot \tag{7}$$

It is difficult to make any accurate estimate of Δ from the anatomy of the cochlea but the following consideration may help to understand the factors involved.

Consider a tube of liquid of length $2x$ and having a cross $w.\delta x$ moving slowly to and fro in a larger tube filled with the same liquid. This is illustrated in cross section by Fig. 6. When the cross sections of such tubes are small the frictional force is proportional to the interacting surface namely $2x.\ 2(w+\delta x)$, multiplied by the velocity, the factor of proportionality being η/d where η is the coefficient of viscosity, and d the effective distance of the surface from the containing walls.

The value of d is approximately that shown in Fig. 6. The damping factor for such a vibrating column when δx is small compared to w, will then be approximately

$$\Delta = .868\frac{\eta}{\delta x . d} \cdot \tag{8}$$

It is seen that Δ is independent of the length of the column but depends upon its thickness δx. If the thickness is just sufficient to cover a rod of Corti then $\Delta = 30$ and if it is equal to w then $\Delta = 570$ bels per second.

It would be expected that the value of Δ for the case of the element of the basilar membrane considered above would be between these limits. Also this type of consideration leads one to suppose that Δ is

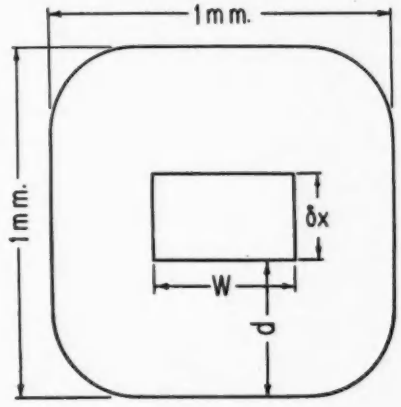


FIG. 6

independent of x, that is, it is the same for different positions along the basilar membrane.

Using the values given for F, m and s we can now write down the expression for the displacement y of the element δx from its position of equilibrium

$$y = A \cos(\omega t - \Theta) \tag{9}$$

where the amplitude A is given by

$$A = \frac{1}{4\pi^2}\frac{kf}{k_m}\frac{P}{x[(f_0^2 - f)^2 + (.37f\Delta)^2]^{1/2}} \tag{10}$$

and the phase Θ by

$$\tan\Theta = .37\Delta\frac{f}{f_0^2 - f^2} \cdot \tag{11}$$

It was seen that factor k_f varied from zero for very low frequencies to unity for high frequencies and that k_m had a value of approximately 2 at least if the motion were similar to that indicated in Fig. 6. Taking the value of k_f equal to unity, k_m equal to 2, and $\Delta = 70$ and $P = 1$ bar, calculations of A and Θ were made. They are shown in Table I. The relation between f_0 and x used was that given in equation (6).

TABLE I

$f=75$			$f=125$			$f=250$			$f=500$		
x	$A\times10^8$	Θ	x	$A\times10^8$	Θ	x	$A\times10^8$	Θ	x	$A\times10^8$	Θ
10	.2	0	10	.2	0°	10	.2	0	10	.2	0
20	2	0	20	2	1°	20	3	2°	15	.8	1°
25	7	2°	25	8	8°	25	30	27°	20	8	9
30	28	7°	30	71	33°	25.8	67	62°	20.5	16	19°
31	40	11°	31	126	90°	26	75	90°	21	46	90°
						27	28	157°	21.5	17	158°
						31	8	172°	22	9	168°
									25	3	176°
									31		

$f=1000$			$f=2000$			$f=4000$		
x	$A\times10^8$	Θ	x	$A\times10^8$	Θ	x	$A\times10^8$	Θ
10	.3	0°	5	.2	0°	5	.5	1°
15	2.6	5°	10	1.0	2°	5.5	1.0	2°
15.5	5.4	10°	10.5	2.0	5°	5.8	2.3	6°
16.0	30.4	90°	11.0	22	90°	6.0	20	90°
16.5	5.8	169°	11.5	2.3	174°	6.2	2.4	173°
17.0	3.1	174°	12.0	1.1	177	6.5	1.0	177°
20.0	.9	178°	15.0	.3	179°	7.0	.7	178°
31	.4	178°	31	.1	179°	20	.04	180°

The values of A are given in 10^{-8} cm which is the order of magnitude for the diameter of the molecules. If values of velocity or acceleration are calculated for any one frequency the relative values will be the same as those shown in the table. Even the bending of the membrane will vary sinusoidally with amplitudes which are not greatly different from those shown. This suggests that the stimulation pattern for a tone without harmonics (either objective or subjective) will be the same as the amplitude pattern.

It will be seen that the resonant peaks are very much sharper for the high frequencies than for the low ones, a fact which is in agreement with the data on masking as will be seen.

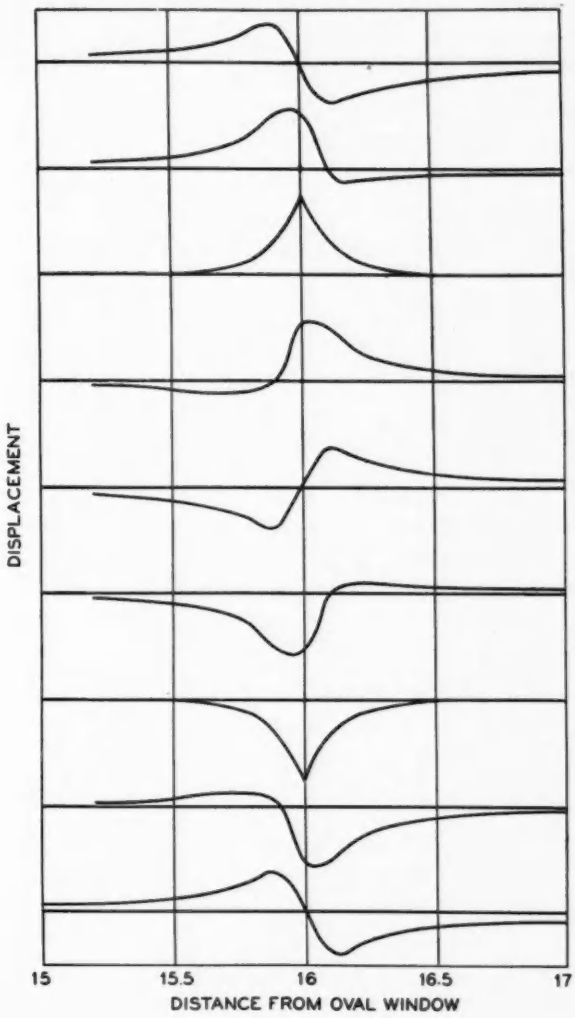


FIG. 7

It is important to notice that, according to these assumptions and most any other reasonable assumption one can make, the part of the membrane next to the oval window leads in the phase of vibration, so that if motion pictures were taken with exposures $T/8$ seconds apart where T is the period for one complete cycle, we would get a series of

pictures like those shown in Fig. 7 which are plotted from the results in Table I for the case of 1000 cycles.

Bekesy[7] has pointed out that such a motion of the membrane would tend to produce whirls in the liquid above the position of maximum stimulation. When the exciting tones were loud, he observed such whirls both in his models of the ear and in dissected human ears. He considers that it is the steady pressure against the membrane produced by this whirl that causes the nerve stimulation. He points out that such a whirl would cause a movement of the liquid in the semi-circular canals so as to effect the sense of balance and cause an observer exposed to such a loud tone of high pitch to tilt his head toward the ear receiving the sound. He claims that such an effect has been noted a sufficient number of times to substantiate this point of view. It seems doubtful, however, that such a whirl would be set up except for very loud tones. It also seems difficult to explain masking data on any assumption other than that the stimulation of the nerve endings is due to the vibration of the basilar membrane which moves in some way similar to that described. The mechanism of this stimulation will be dealt with later.

TABLE II

f	Relative Reciprocal Threshold Pressures	Relative Displacements	Relative Velocities	Relative Accelerations
75	.004	1.22	.09	.007
125	.024	4.14	.52	.065
250	.125	2.47	.62	.17
500	.500	1.52	.76	.38
1000	1.000	1.00	1.00	1.00
2000	1.250	.73	1.47	2.84
4000	1.250	.62	2.48	9.92
8000	.180	.64 .027	5.12 .22	40.96 1.72
12000	.050	.0036	.043	.52

In Table II the relative values of the maximum displacement, velocity, and acceleration of impressed tones of various frequencies f, as calculated above, and also the reciprocal values of the pressure change in the air necessary to make the tone just audible are given. The second value at 8000 was calculated on the assumption that the nerve

[7] Phys. Zeit. Vol. 29, pp. 793–810, Nov. 15, 1928.

endings stop at $x = 1.5$ instead of at $x = 1.0$ mm. It is seen that for frequencies above 1000 cycles the threshold values agree better with the maximum velocity values and for frequencies below 1000 cycles better with the maximum acceleration values. But these calculations were made on the assumption that P, the pressure at the oval window, was constant for tones of different frequencies. As was pointed out above, the results of impedance measurements[8] upon the ear indicate that for constant intensity tones in the air having frequencies below 1000 cycles, a constant amplitude rather than a constant velocity of the ear drum is produced. Consequently, the value of P will be proportional to the frequency for such tones in this frequency range, assuming, of course, that no other change takes place in the process of transmission through the middle ear. Introducing this new factor into the calculation, the velocity values in this frequency range will be the same as those given under the heading "Relative Accelerations" in Table II. With this correction then, the velocity values agree with as good an accuracy as could possibly be expected from such an analysis with the reciprocal threshold values throughout the entire range. The factor k_f becomes smaller for the low frequencies which likewise tends to make the velocities smaller, especially those of very low frequencies. Although this evidence is far from conclusive, it is evidence for supposing that equal stimulations will be produced at different parts of the basilar membrane which have equal velocities even though the frequencies are different. For obvious reasons it would not be expected that Equations (10) and (11) would give values which would agree with the stimulation space pattern except for small distances on either side of the maximum stimulation position.

Stimulation Patterns Determined from Masking Data

These space patterns for the entire length of the membrane, however, can be obtained from the masking data. Such data taken on my left ear by the method described by Wegel and Lane[9] are given in Figs. 8 and 9 for stimulating tones having frequencies 75, 125, 250, 500 and 1000. The intensity level of the tones in the ear were adjusted to be equal to -20 *db*. To do this the sensation levels were made equal to 27, 40, 54, 65 and 72 *db* as indicated in the figures. The vertical lines represent the sensation levels of the subjective harmonics as deter-

[8] See Tröger's results page 34 of paper already cited.

[9] Phys. Rev. 19, 492, May, 1922.

mined by the "best beat" method. The ordinates of the curves give the masking, that is, the number of decibels that the threshold of audibility is shifted due to the presence of the stimulating tone. It is interesting to note that under these conditions the same number of

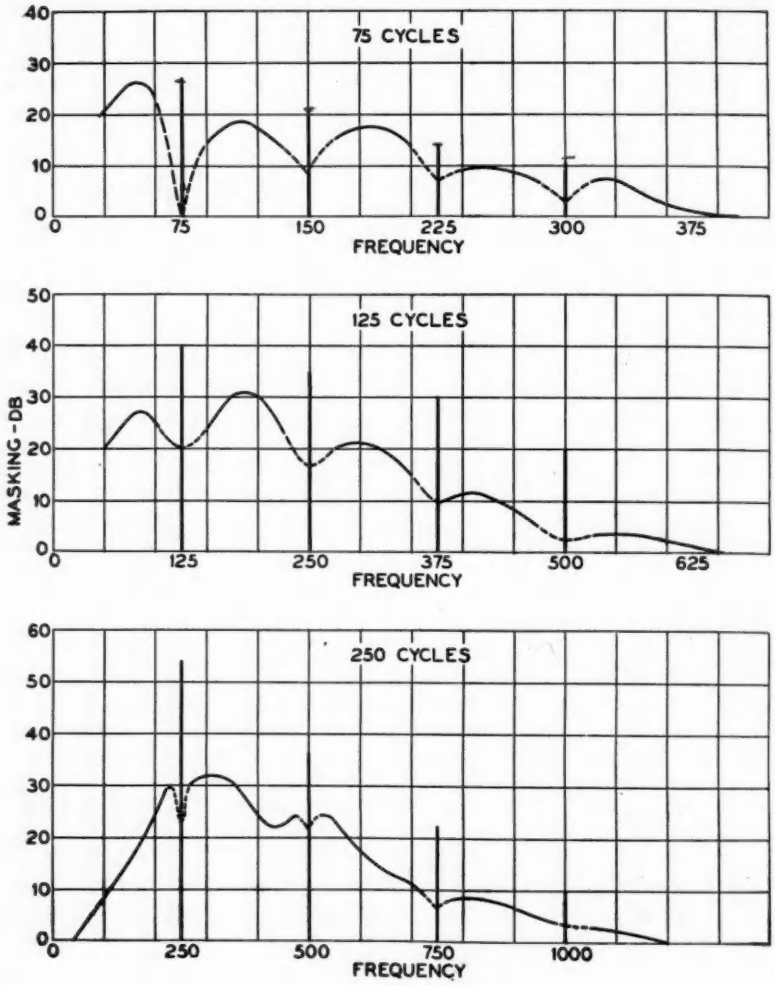


FIG. 8

audible subjective harmonics are created by each of the five stimulating tones and with approximately the same intensity levels. For example, the intensity levels of the second harmonic in each case are 42, 41, 49, 34 and 42 decibels below zero level. Similarly, the third harmonics are

56, 52, 67, 60, 53 and the fourth harmonics are 67, 75, 82, 85 and 86 decibels below zero level.

For the 75 cycle tone and for tones of lower pitch subjective harmonics could be detected as long as the tone could be heard. For

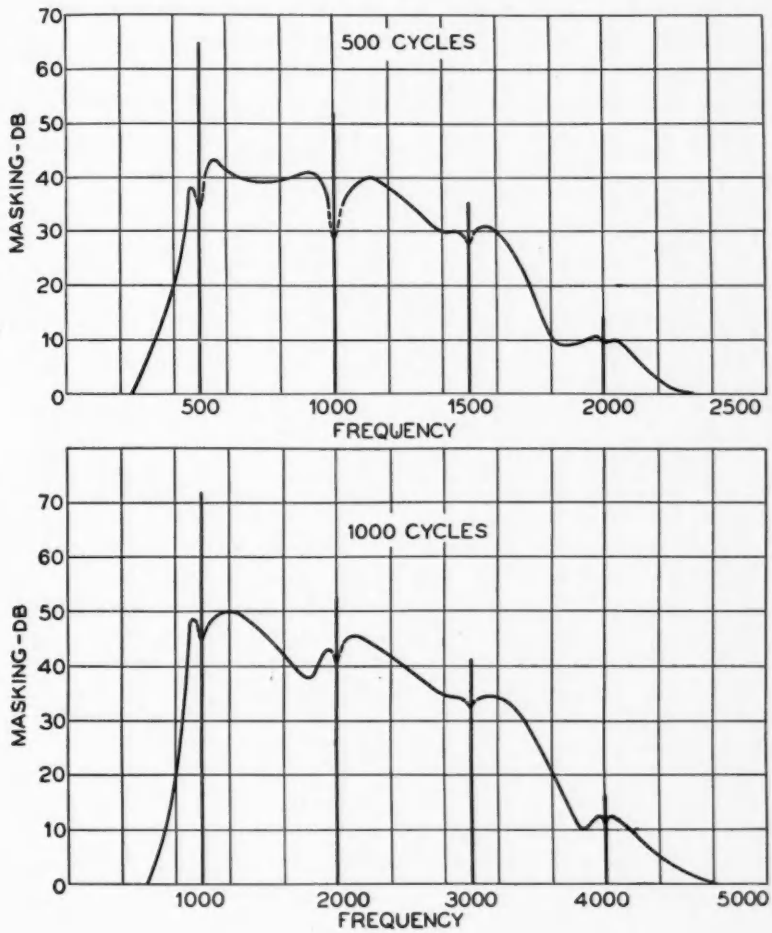


FIG. 9

example, if tones having frequencies of 50 and 103 are applied to the ear simultaneously, then for any intensity of the 50 cycle tone that can be heard, an intensity of the 103 cycle tone can be found so that the sensation of 3 beats per second will be produced. This fact makes it difficult to use masking data for determing the form of vibration of

the basilar membrane for a pure low pitched tone being impressed at the oval window of the cochlea.

If now we use the relation between f and x given by Equation (10), we can estimate stimulation level curves from these masking curves. The level for the position of the harmonics will be directly that given in Figs. 8 and 9. For intermediate positions where only masking takes place, the procedure is more uncertain. If we assume that the masked tone is just perceptible when it is at a level corresponding to that necessary when the masking and masked tone have the same frequency

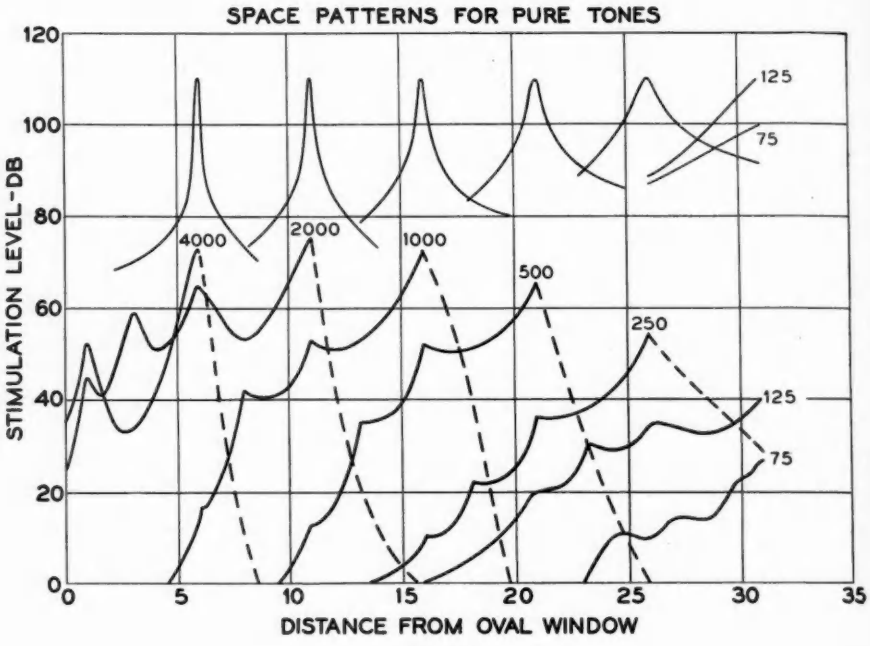


FIG. 10

then the data on minimum perceptible differences in intensity can be applied. These data indicate that when the masking tone has a greater sensation level than 40 *db* then the masked tone is always approximately at a level which is 20 *db* lower than the level of the masking tone. As the sensation level of the masking tone decreases below 40 *db* the difference between the levels of the masking tone and the masked tone decreases from 20 *db* to zero. In this way the curves given in Fig. 10 were obtained. The right side of each curve is shown dotted so that the curve might be followed more easily. The curves indicated by the light lines at the top of the figure are the calculated ones taken from Table I,

the amplitude values being transformed to decibels and the peak values adjusted to be at the same level. It is seen that for a range of 3 or 4 mm. along the basilar membrane, these curves have the same general shape as the masking data curves. For greater range the latter curves fall off much more rapidly than the simple resonance curves. The curves for 2000 cycles and 4000 cycles were taken from data obtained by Mr. F. H. Graham.

It is seen then that when each of the five tones considered above is impressed upon the ear, space patterns similar to those shown in Fig. 10

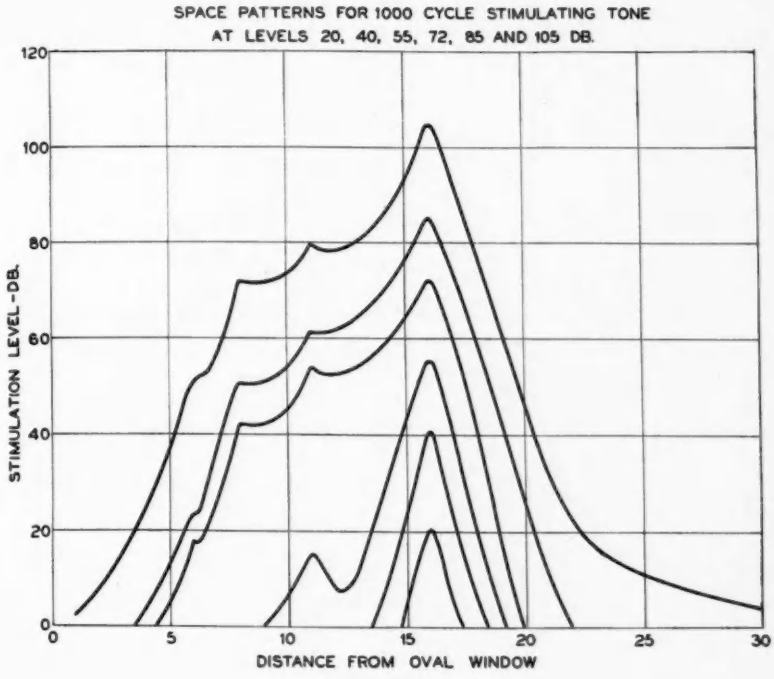


FIG. 11

are produced corresponding to the sinusoidal time pattern in the air.

Similarly, for every sound reaching the ear a space stimulation pattern corresponding to the time pattern of the sound in the air is produced on the basilar membrane. This space pattern is transferred to the brain by a nerve cable containing about 3000 individual nerve fibres. This space pattern is dependent both upon the kind of sound and also the intensity with which it is applied to the ear. This is illustrated by the stimulation patterns shown in Fig. 11 for a 1000 cycle tone impressed with the various intensities indicated by the peaks of the

curves. These curves were estimated from data on best beats and masking as described above.

MECHANISM OF NERVE STIMULATION

Before considering how these stimulation patterns are related to loudness, let us consider more closely the mechanism of stimulating the nerves. Some of the nerve endings will be very sensitive while others will be very insensitive; others will have degrees of sensitivity scattered between these extremes. Let us take the stimulation level at a point on the basilar membrane as the number of bels that the velocity at that point is above the agitation of the membrane which would just stimulate the most sensitive fibre in a group situated along one millimeter of length of basilar membrane. Then if the nerve endings in the ear have properties similar to those in the eye the distribution of threshold values will be as follows. Let z be the fraction of fibres which have threshold values below the stimulation level β, then

$$\beta - \beta_0 = \log \frac{z^2}{1 - z} \cdot \tag{12}$$

The constant β_0 is the value of the right-hand member at the threshold. As shown by Hecht[10] this relation holds for the nerve endings in the eye. The constant β_0 depends upon the value of z at the threshold.

As the stimulation level goes above the threshold of a nerve ending the firing of nerve impulses from that ending becomes more rapid. Let us consider this mechanism for a moment. We will assume that there are two opposing operations in the process of stimulating the nerves. The first, which may be called the active process, is dependent upon the magnitude of the motion and tends to create either a substance or energy which acts like a catalytic agent and produces a nerve discharge. The second, which may be called the reactive process, tends to destroy or annul the active process. For example, let us call the thing being created the discharger which may be the energy of agitation of the element of the nerve ending or a fluid produced by some chemical process or some kind of electrical accumulation. The active process will tend to create this discharger at a rate depending upon the stimulating force and the reactive process will destroy the discharger at a rate depending upon the amount present. If the discharger is the amount of

[10] Light Sensitivity of Animals. Proceedings of the Michelson Mtg. of the Optical Society of Amer. Nov., 1928.

energy of vibration, then the action of frictional forces furnishes the reactive process. Let us call the amount of discharger present q. As a first approximation it will be assumed that the rate at which q is created is proportional to the energy[11] of agitation and the rate at which it is destroyed is proportional to q. Formulating these assumptions into a differential equation

$$\frac{dq}{dt} = av^2 - bq \tag{13}$$

where a and b are quantities independent of x or t and v is the velocity of the membrane where the nerve ending is situated.

For a periodic force such as that produced by a pure tone

$$v = V \cos \omega t. \tag{14}$$

Substituting this in Equation (13) and solving under the conditions

$$q = 0 \quad \text{when} \quad t = 0$$

$$q = \frac{aV^2}{2b}\left(1 - e^{-bt} + \frac{1}{\sqrt{1 + \left(\frac{2\omega}{b}\right)^2}}\left[\cos(2\omega t - \Theta) - e^{-bt}\cos\Theta\right]\right) \tag{15}$$

where $\tan \Theta = 2\omega/b$. If b is small compared to ω then

$$q = \frac{aV^2}{2b}\left(1 - e^{-bt} - \frac{b}{2\omega}\sin 2\omega t\right). \tag{16}$$

The amount of discharger present is proportional to the energy of agitation, inversely proportional to b, and directly proportional to the time factor in the parentheses. This process continues until the threshold value of the nerve ending is reached when a new process starts; a nerve discharge then takes place. While this second process is going on the equilibrum is upset so that the creation of the discharger stops. This period of the stopping of the first process is the reaction time τ of the nerve ending. After the reaction interval, the active process starts again. At the high intensities the number of discharges is governed mostly by the reaction time while at the low intensities mainly by the time for the discharger to accumulate.

[11] This assumption is arbitrary and results in predicting that the double frequency will occur in the brain time pattern. This is used only for illustrative purposes. Other assumptions can be made which will result in leaving the fundamental and its harmonics in the time pattern of the brain.

To illustrate the form of this equation, the graph of the time function in the brackets is shown in Fig. 12 when $b=125$ and the frequencies of 125 and 500 cycles per second. Any fibre may obtain the requisite amount of q for a discharge either by having a large velocity V and a small time t or a small velocity V and long time t. Consequently, for any given value of V the sensitive fibres will require only a short time to discharge and the insensitive ones a long time. It is seen that for the 500 cycle tone q increases in the interval .001 to .003 second about 10 fold while in the same length interval .003 to .005 it remains practically constant. It is evident then that the number of discharges coming from a stimulated patch of nerve fibres which take place in the first interval

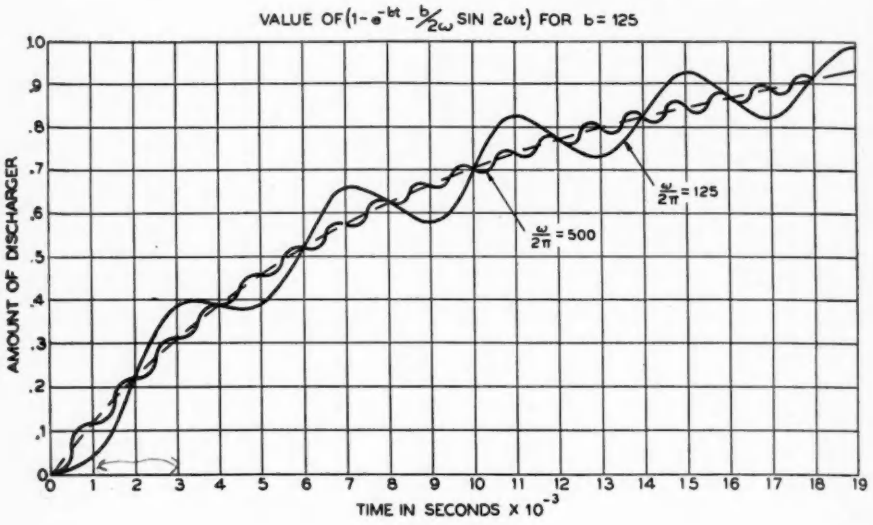


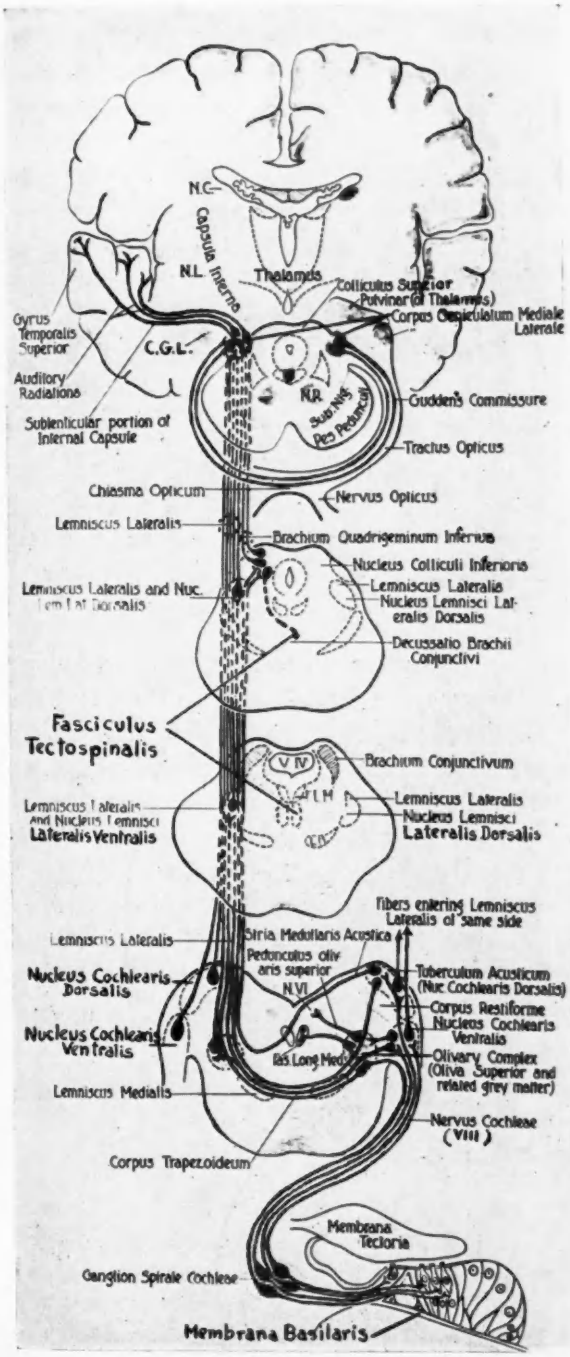
FIG. 12

will be very much greater than in the second interval. Consequently, the bombardment of nerve discharges reaching the brain will have times of maximum intensity which are separated by intervals equal to the half period of the oscillating body producing the stimulating sound. In other words, a time pattern of the changes taking place in the medium transmitting the sound wave is transmitted to the brain. In passing to the brain it follows the nerve tract shown in Fig. 13.

Although the nerve fibres are interrupted by synapse as indicated, where a restimulation is necessary, it seems reasonable to expect some sort of correlation between the rate that the nerve discharges are being received and the time pattern of the original wave. As the frequency

AUDITORY TRACT

According to A. R. Rasmussen, Professor of Neurology at the University of Minnesota



Note: You are looking down on transverse sections through various levels of the brain except in case of the upper part of the upper level (see below). The dorsal side of the brain is up.

Only one lateral lemniscus is carried up from level to level. If the fibers from the cochlear nuclei of the right-hand side were carried up they would follow a course similar to that taken by the fibers that cross to the left-hand side. The brain is bilaterally symmetrical.

The lower level, which is through the extreme lower part of the pons, is slightly oblique, being a little lower dorsally than ventrally so as to intersect the dorsal cochlear nucleus.

The second level (from below, not counting the drawing of the organ of Corti) is through the middle of the pons. The third level is through the upper part of the pons ventrally and lower part of the mid-brain dorsally. The lower part of the upper level shown is a cross-section through the upper part of the mid-brain. In order to show the position of the auditory radiations, it was necessary to resort to a vertical section through the cerebral hemispheres. This vertical section is so placed as to intersect the cross-section level of the upper mid-brain in the region of the medial geniculate body.

FIG. 13

of the stimulating tone becomes higher the time pattern becomes less distinct since the ratio $b/2\omega$ becomes smaller. The case for 500 cycles is also shown in Fig. 12. The variation from the exponential lines for a 5000 cycle tone would be 1/10 that shown for the 500 cycle tone.

For a large value of V the nerve will discharge in a very short time and for small values it will take a longer time and may not discharge at all. Let V_1 be a value corresponding to a stimulation level β such that q will be at the threshold for a certain fibre when acted upon for a time long enough that e^{-bt} is negligible compared to unity, that is, so that the time factor in Equation (16) has reached its maximum value of $(1+b/2\omega)$. Let V_2 be any other velocity above V_1 corresponding to a level α which will give the same value of q in a time t. Then

$$V_1^2\left(1+\frac{b}{2\omega}\right)=V_2^2\left(1-e^{-bt}-\frac{b}{2\omega}\sin 2\omega t\right) \tag{17}$$

$$\alpha-\beta=\log\frac{1+\dfrac{b}{2\omega}}{1-e^{-bt}-\dfrac{b}{2\omega}\sin\omega t}\cdot\log \tag{18}$$

The value t in this expression is the time necessary for a nerve ending to be acted upon at a level which is $\alpha-\beta$ bels above its threshold level before it starts a nerve discharge. As mentioned above, the reaction time τ is the time necessary for the nerve to discharge and recuperate, thus being ready again for another stimulation. Hence the time $t+\tau$ is time interval before the nerve is ready to start the process over again. At the end of this time interval the phase of vibration will not in general be the same as in the beginning. However, on the average, we can neglect the term $b/2\omega \sin 2\omega t$ when determining at each level the average rate of firing since it simply has the effect of making the time interval oscillate on either side of the value obtained when this term is omitted. The rate so determined can then be applied to all frequencies and it depends only upon the stimulation level. With this understanding the average rate of firing from a nerve which is stimulated at a level $(\alpha-\beta)$ bels above its threshold is given by

$$r=\frac{1}{t+\tau}. \tag{19}$$

The fraction of threshold values lying between the levels β and $\beta+d\beta$ is $(dz/d\beta)d\beta$. Consequently, if n is number of fibres per mm. the number of nerve discharges per second coming from a small length dx of basilar membrane will be given by the expression

$$ndx\int_0^\alpha r\cdot\frac{dz}{d\beta}d\beta. \tag{20}$$

For convenience let

$$S(\alpha)=\int_0^\alpha r\cdot\frac{dz}{d\beta}d\beta. \tag{21}$$

The value of $dz/d\beta$ can be obtained from Equation (12) and the value of r from (18) and (19). The total rate of discharge R going to the brain is then

$$R=\int_0^{31} nS(\alpha)dx. \tag{22}$$

Although the values of these integrals cannot be obtained analytically, they can be readily evaluated by graphical methods. To find the value of $S(\alpha)$ change the variable from β to z then

$$S(\alpha)=\int_z^z \cdot r\cdot dz \tag{23}$$

where z_0 corresponds to the threshold value, that is, the fraction of fibres that must be excited to produce a sensation. Taking $\tau=.002$ second and $b=125$, and $z_0=10^{-4}$ values of $S(\alpha)$ were calculated as shown in Fig. 14. The ordinates give values of $S(\alpha)$ which is a number that when multiplied by the number of nerve endings at the stimulation level α will give the total discharges per second from such nerve endings. There are about 1000 rods of Corti per millimeter along the basilar membrane. Then, according to this calculation, there would be 10 discharges per second from a millimeter at the threshold or zero stimulation level and 500,000 discharges per second when stimulated for its maximum discharge rate.

Comparison of the Space-Time Pattern Theory with the Experimental Facts of Audition

It was seen that variation in ear sensitivity with frequency was satisfactorily explained by the variation of the impedance of the ear,

and variation of velocity of basilar membrane due to the mechanical action of the cochlea.

There is no doubt that loudness is closely associated with the total discharges reaching the brain. It is seen by Fig. 14 that the possible range of loudness from a single mm. of membrane vibrating at different amplitudes is 11 or 12 bels. If we add to the one millimeter the possible 30 millimeters, the total impulses will be multiplied by 30 which is equivalent to increasing the level 2 or 3 bels in the high intensity regions. It is thus seen that this mechanism provides for ample range of the

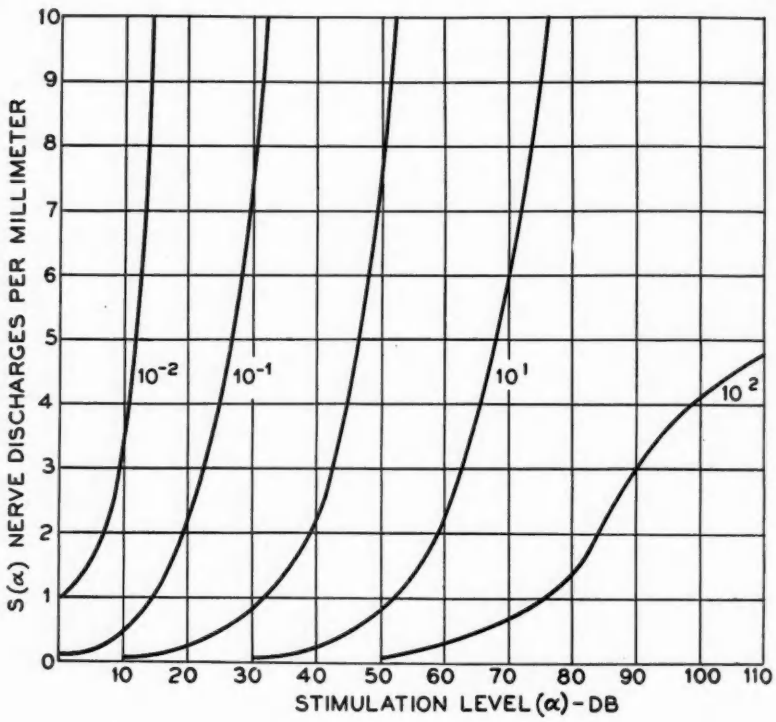


FIG. 14

loudness magnitudes used in audition. When a tone is gradually increased in intensity from the threshold, it first stimulates a small patch; this grows in length and then other patches corresponding to the subjective harmonics are started until finally some nerve fibres along the entire length are stimulated. If the membrane vibrated the same for all tones, similar for example to the diaphragm of the Wente condenser transmitter, then the loudness for all tones should be the same

for the same sensation level which is contrary to the observed facts. It is the difference in form of vibration of the basilar membrane that accounts for the difference in loudness of tones having the same sensation level.

The functions calculated above have been based on the assumption that as the intensity of stimulation increases the rate of discharging the nerve also increases until the limit of 500 per second is reached. Experiments on nerves have shown that as the nerve is continually stimulated it is fatigued, after which the maximum rate is very much reduced. Consequently, if a tone of given frequency stimulates the ear for some time then immediately after such stimulation the loudness of tones in the same frequency region will be reduced. This effect has been recently measured by Bekesy.[12] His results indicate that changes in loudness as much as 30 *db* occur imediately after strong stimulations.

Similarly, the threshold of audibility should be shifted if taken immediately after such strong simulations. Measurements in our laboratory, which will soon be published, indicate that especially for low pitched tones such shifts amount to as much as 25 *db* and some shifts take place as long as 2 minutes after the stimulating tone is stopped. So the relations developed above can be applied only when the nerves are not fatigued. Both of these effects show that the changes are greatest for the tones near the pitch of the stimulating tone and decrease to zero for tones far removed from the pitch of the stimulating tone. This then is additional evidence that certain regions only along the basilar membrane are stimulated by a simple tone.

TABLE III

Frequency	Sensation Level	*R*	Calculated Loudness	Observed Loudness			
				F	*Ror*	*Re*	*G*
75	27 *db*	1,000	35	36	50	40	33
125	40	10,000	50	50	57		
250	54	40,000	61	60	62		
500	65	108,000	69	72	66		
2000	75	250,000	77	79	76		
4000	73	140,000	71	70			

It has been seen that due to the introduction of more subjective harmonics and also due to the form of vibration having less sharp resonant

[12] Phys. Zeit. Vol. 30, No. 21, Nov., 1929.

peaks, the region of maximum stimulation is much broader for tones of low frequency than for those of high frequency when both have the same sensation level. For this reason, these low tones will send more nerve impulses to the brain and consequently will be louder.

The values of R of Equation (22) were calculated for the seven tones shown in Fig. 10, taking n equal to 1000 per millimeter. The results are shown in the third column of Table III. Similarly, the values of R for a 1000 cycle tone at various levels were calculated from the curves in Fig. 11. The results are given by the curves in Fig. 15.

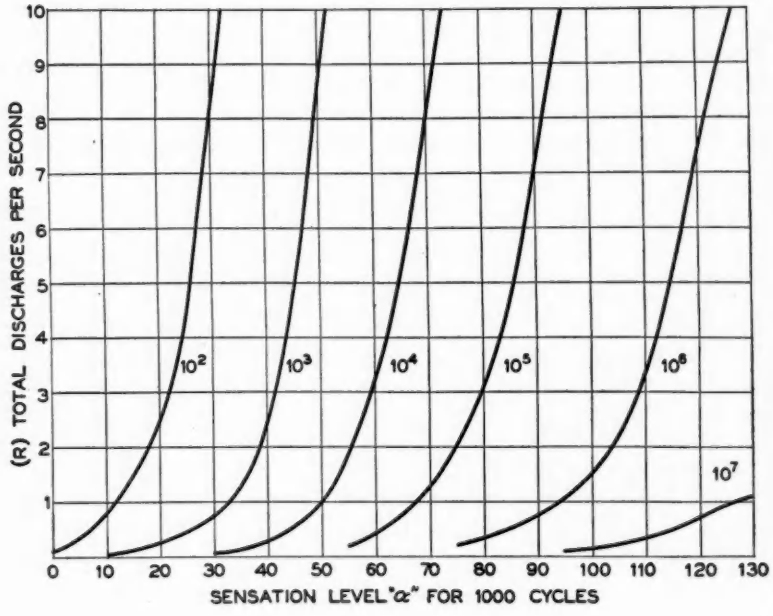


FIG. 15

The loudness of a sound is defined[13] as the sensation level of a 1000 cycle tone which is adjusted to sound equally loud. Consequently, the loudness numbers of each of the tones whose frequencies are shown in column (1) of Table III can be obtained by comparing the values of R with those in Fig. 15 and noting the corresponding sensation level. The values in the fourth column were obtained in this way. Inasmuch as all the data for constructing these curves were based upon measurements on my left ear, some loudness measurements were also made with this

[13] See Book "Speech & Hearing" page 226.

ear. The level of a 1000 cycle tone was adjusted so that it sounded equally loud to the six tones all introduced at the common intensity level of -20 *db*. The observer listened to the two tones alternately so that the fatigue effect should be eliminated. The observed results are shown in the fifth column. Inasmuch as these values were somewhat different than those reported by Kingsbury, I thought my judgment may have been influenced by knowing what to expect, so I asked another observer not familiar with the subject to make similar balances. His results are shown in the sixth column. Inasmuch as the difference was rather large for the 75 cycle tone, two other observers were asked to make balances for this tone. Their results are shown in the seventh and eighth columns. The agreement between the observed and calculated values seems to agree within the limits of observational error although the observed values tend to be higher for the two tones of lowest pitch. It is possible that the phenomenon of "volume" influences the observer to judge the low tones louder than otherwise would be the case. This may be one of the elements that enters into a judgment of equal loudness.

Let us now see how the differential sensitivity of the ear for pitch and intensity fits into the picture. These two effects are closely related for in each case it is a small change of the vibration form of the basilar membrane that can be detected, at least for the higher frequencies. First consider small changes in pitch.

Since the peaks are sharper for the high pitched tones, it would be expected that smaller changes in pitch could be detected for these tones which is in accordance with the facts. It would be expected also that smaller differences could be detected at the high intensities for under these conditions the several peaks due to the subjective harmonics will aid in making such detections. This also agrees with the observed facts. The smallest pitch change that can be detected is about .25 centi-octaves which corresponds to a shift in the stimulation peak of about 1/100 millimeter covering about 10 nerve endings. So this effect is not inconsistent with the anatomy of the ear.

The smallest intensity level change that can be detected is about .25 *db*. This corresponds to about a 5% change in R. There seems to be no reason why this should or should not change with intensity level or pitch unless it is a change in the space pattern that is detected rather than a change in R. When two tones of the same frequency are compared it is probably a change in the space pattern that enables one to

detect the difference in intensity. If the change in form of the space pattern is the dominating influence for such small changes in intensity, then perceptible changes in intensity should be influenced by the same factors as those mentioned for perceptible pitch changes. This latter view seems to agree with the experimental facts concerning differential sensitivity for intensity.

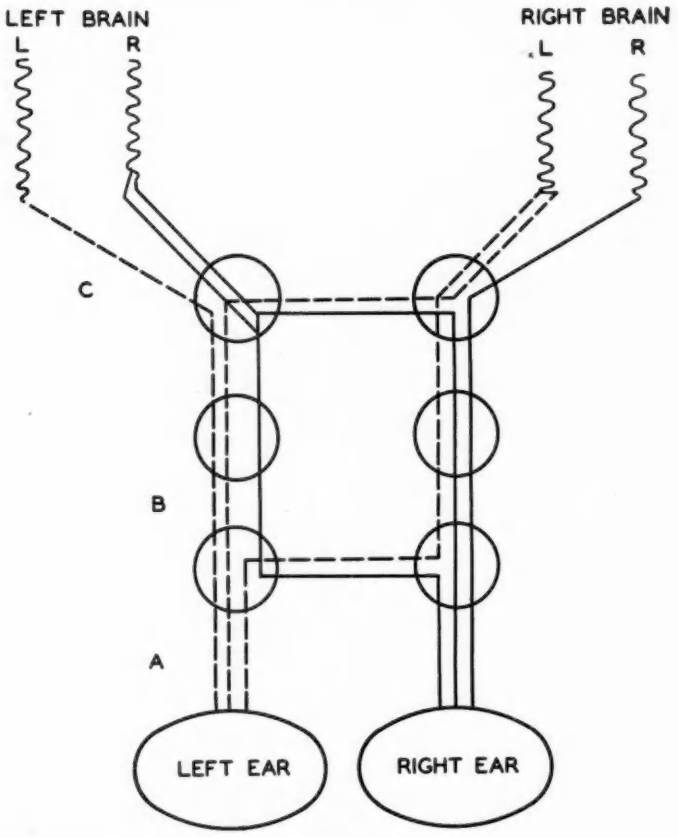


FIG. 16

It also agrees with the experimental fact that it is very difficult to compare the loudness of tones of different frequency. In such a case the relative magnitudes of R must be judged.

There then remains the facts of binaural audition and some of the more subtle psychological effects, such as "volume" and "brightness."

In Fig. 16 a diagrammatic sketch of the auditory tract taken from Fig. 13 is shown. The circle represent places where some nerve fibres

stop and others begin. These are scattered along the length of the nerve path. Some nerves have a continuous passage from the cochlea to the auditory brain center, while others may be interrupted three or four times by such synapse. As shown the nerves running from each ear are cross-connected at two points. More nerve fibres coming from the left ear terminate in the right side of the brain and vice versa.

So we see that according to the conceptions given above, when the left ear receives a tone then there should be two space patterns in the brain which are almost alike, one on the left side and one on the right. They are designated by the letter *L*. Similarly, a sound in the right ear produces the two patterns labelled *R*. If the nerve is interrupted at *A*, the left ear will be totally deaf. If, however, it is interrupted at *B* or *C*, only a slight diminution in hearing in either ear will occur. Cases of brain tumor have been found where it was necessary to remove one of the auditory brain centers and also a considerable portion around it and still the patient recovered. Although it left one side of the body paralyzed, hearing tests on such a person indicated that the two ears had practically the same sensitivity as before the operation.

Although some of the nerve fibres from the two ears are placed near together for part of their length and terminate close together in the periphery of the brain, there is little interference between them. This is shown by the experimental fact that a tone introduced in one ear will produce little masking to a sound introduced into the other one. Wegel and Lane showed that a tone to be masked in an opposite ear to that receiving the masking tone needed to be 1,000,000 more intense, that is, at a 60 *db* higher intensity level than if introduced in the same ear as the masking tone.

Also, binaural "objective" beats will not occur until the intensity of tone in one ear is at least 60 *db* above that in the other ear. Objective beats are distinguished from "subjective" beats by the following criteria. They can be readily heard by every observer and can be produced at all audible frequencies while the "subjective" beats can ony be heard by about 80% of observers and only then when the frequencies are below 1000 cycles per second. The objective beats are undoubtedly produced by the more intense sound forcing the bones of the head into vibration so that the sound is conducted to the cochlea of the opposite ear. Consequently, such an intense tone produces the same effect as if it were attenuated and introduced directly into the same ear as the other tone.

The patterns from the two ears are laid down close in duplicate so that an accurate comparison can be made. Since the times of maximum stimulation have been shown to be directly correlated to the times of maximum displacement in the medium conducting the sound, the differences of phase which occur at the two ears will manifest themselves as phase differences in the times of maximum stimulation between *L* and *R* on both sides of the brain. In locating the direction of a source of sound both the differences in intensity and also the differences in phase are used. Due to a process of education we have come to associate a definite direction of sound with this difference in the intensity of the stimulation between *L* and *R* and the times between the occurrence of maximum stimulation. It has been seen that the definiteness of this time pattern in the brain decreases with increasing frequency which agrees with the fact that the location of the direction of a source of sound having frequencies higher than 1000 cycles is extremely difficult.

If tones having the same frequency and intensity are applied to the two ears and then the phase changed, it causes a sound image to rotate around the head. Such an effect is related to our experience of locating sounds but it is a new experience since the usual change in intensity is missing, but most persons can make an adjustment for this absence of intensity change and assign a definite direction for the sound image. The changing of phase can be produced by making the tones in the two ears slightly different in frequency. Then the phenomenon known as "subjective" binaural beats is produced. As mentioned above, these beats cannot be heard for tones having high frequencies which is in accordance with the theory outlined.

SUMMARY

Summarizing then, the pitch of a tone is determined both by the position of its maximum stimulation on the basilar membrane and also by the time pattern sent to the brain. The former is probably more important for the high tones and the latter for the low tones. The loudness is dependent upon the number of nerve impulses per second reaching the brain and possibly somewhat upon the extent of the stimulated patch. The experience called by psychologists "volume" or "extension" is no doubt identified with the length of the stimulated patch on the basilar membrane. This extension is carried to the brain and forms a portion of excited brain matter of a definite size. It is then this size that determines our sensation of the "volume" of a tone. The low pitched or

complex tones have a large "volume" while the high pitched tones have a small one.

The psychological experience called "brightness" may be identified with the sharpness of the peaks in the vibration form of the basilar membrane as suggested by Dr. Troland. The high tones give the sense of brightness while the low tones the sense of dullness which as seen from Fig. 10 corresponds to the sharpness of the peaks in the vibration form.

The time pattern in the air is converted into a space pattern on the basilar membrane. The nerve endings are excited in such a way that this space pattern is transferred to the brain and produces two similar space patterns in the brain, one on the left and the other on the right side. Enough of the time pattern in the air is sent to each of these stimulated patches to make times of maximum stimulation in each patch detectable. So when listening to a sound with both ears, there are four space patterns in the brain produced, each carrying also some sort of time pattern. It is a recognition of the changes in these patterns that accounts for all the phenomena of audition.

BINAURAL HEARING[1]

By G. W. Stewart
University of Iowa

Any adequate theory of hearing must explain all the perceptions of binaural audition. I shall speak of three important divisions in reference to these binaural effects, not because they are new to those present, but because they give certain facts which must be explained in any really satisfactory theory of hearing.

The first is the *binaural intensity effect.* Here there are two significant facts. By different intensities at the ears, the frequency and phase conditions being alike at the two ears, there will be caused different displacements of the phantom source of sound from the median plane, the angle of displacement being proportional to the logarithm of the ratio of the intensities at the two ears. I have tested this law quantitatively for several frequencies from 256 to 1000 cycles and have found it to hold admirably. I have also shown quantitatively that the intensity ratio required for a given angular displacement of the phantom source is many times greater than the difference of intensity that would be experienced at the ears were such a source actually placed at the given angular displacement from the median plane and not unusually close to the head. This statement refers to frequencies say less than 1200. Then, at these frequencies, the intensity effect cannot be the cause of localization. Later the cause will be assigned. But for the moment your attention is directed to the quantitative fact which has not been previously suggested, that this logarithmic law cannot have its origin in causes produced by the experiences of the individual, for, in the frequency region named, he has no correlation between intensity and localization as perceived. This logarithmic law is then only a part of something, I know not what, that is more significant. An adequate theory of hearing must then explain not only the cause for the logarithmic intensity effect, but also should state a more profound law.

The second important fact in the binaural intensity effect is that in many individuals there are certain large regions where this effect does not exist. For example, in my own case, for several hundred cycles on either side of 1000, I can increase the intensity at one ear until it is 100 times as great as at the other without at the same time having any

[1] A paper on a special program of the Acoustical Society, December 1929.

motion of the phantom source from the median plane. I have no suggestion in the direction of an explanation of the existence of these lapse regions. That too, must be the burden of an adequate theory of hearing.

The second effect is the *binaural phase effect*. The history of this is a long story. There have been many workers and at the present time there is not entire agreement. My remarks shall be confined to sustained sounds though a great deal has been done with impulsive sounds. The essential fact is that a difference of phase at the ears, the other factors being the same, will produce a localization of the phantom source at an angle from the median plane which is proportional to the phase difference if that is less than 180°. Second, the effect is essentially a difference-of-time effect. This is shown not only by direct experiment but indirectly by the fact that there is no confusion for complex sounds near the median plane. Third, computation shows that at least for frequencies less than approximately 1200 qualitatively the responsibility for localization near the median plane may be placed on the phase effect. The fourth significant point is that *this effect can not be due to intensity either directly or indirectly* because an individual can have a marked phase effect in precisely that frequency region where his intensity effect is actually absent. This discovery is, I fear, lost in the literature because no subsequent workers have referred either to it or to its importance. The evidence is sufficient, however, to cause the removal from psychological treatises of all expression of belief that the phase effect may be directly or indirectly an intensity effect. Regarding the above phase discussion as a whole, it is necessary to conclude that localization in the limited degree here stated depends upon the nature of the stimulus. Turning now to sounds that are not sustained we may remark that psychologists are particularly active in experiments with impulsive sounds. My own inclination would be to understand hearing first with sustained sounds and then extend observations to include impulsive sounds. The latter would be more complicated physically and therefore more difficult to explain satisfactorily. On the other hand, the sustained sounds are more complicated psychologically because of the introduction of fatigue and doubtless other elements. But one is strongly influenced by the greater possibility of quantitative measurements to favor sustained sounds. This difference in views upon the part of psychologists and physicists gives good reason why physicists should be more active in this field. There should now be added to this

brief account the statement that the phase effect, though commonly limited to frequencies less than about 1200, yet with exceptional individuals and some experience may have its limit raised to several thousand at least. The concluding remark concerning the phase effect is merely a reminder that this peculiar and important effect must be explained by any adequate theory of hearing. In this connection we should be reminded that by the perception called localization the ears can detect time differences of the order of 1/100,000 part of a second. The ears seem to be unique in this respect. Whether this remark is very significant or not, I do not know.

The third important binaural effect is that of *binaural beats* or beats occasioned by presenting the two sources one to each ear. These of course demand two tones differing slightly in frequency.

The chief early facts obtained may, with the intensities of the two sounds equal at the ears, be summarized as follows:

If binaural beats occur that are more rapid than one per second there appears one maximum and one minimum in one beat cycle. (According to Lane these binaural beats occur at less than 800 to 1000 cycles.)

When the beat frequency is about one in from one to five seconds, there appear three maxima in one beat cycle, two of them occurring on either side of a phase difference of 180°.

Lane found later an important additional fact: if the intensities are not nearly equal an optimum condition for beats also exists with one intensity 100,000 to 1,000,000 times the other. He explains these quite plausibly by cross conduction. Lane also found a lapse region of intensity ratio where beats do not occur.

All of these facts are not fully explained. I early attempted to explain the equal intensity beats by the excitation of skull sounds and the stimulation of the utricle and saccule. The argument seemed good; at least it showed unquestionably that no cross-conduction theory with two organs of hearing gave a satisfactory explanation. Later Lane set forth the theory that these beats produced by equal intensities were merely subjective, that they depended upon phase equality and the attention of the individual. I confess that both my explanation and Lane's leave me in grave doubt. Both are deficient in explaining the entire phenomena of these maxima both quantitatively and qualitatively. A theory of hearing will doubtless find, in all these facts of binaural beats a test for its adequacy.

I have cited three binaural effects in which there remains much to be explained. In all of these quantitative research only should now be encouraged. Qualitative results are capable of misleading anyone searching for adequate theories. Some may think it is quite unfair to shoulder the desired explanations of these binaural effects upon him who would set up a theory of hearing. But I fear the task must include the biophysics and biochemistry of the entire act from the varying pressure at the drumskin to the perception of the individual. This is a long, long task. No present theory encompasses this range of phenomena. For the immediate future I anticipate that we must find some satisfaction in what are merely contributions to theories of hearing, without adopting any of them as satisfactory.

DISCUSSION
SYMPOSIUM ON HEARING

By Augustus G. Pohlman
Buffalo, New York

Audition is a bio-physical phenomenon and should therefore be discussed from a biological as well as a physical viewpoint. The biologist is particularly interested in the correlation of structure with function. Close study of such correlations, however, often leads to the conclusion that "things are as they are" and sometimes a qualifying clause is added to make the explanation sound more plausible "things are as they are because there is no good reason why they should be otherwise." It is not my purpose to congratulate the essayists on clarity of exposition because clarity of exposition may be nothing more than salesmanship of ideas. I do not believe that the precious pearls of scientific research necessarily come already bored to string on some theory or hypothesis. Accordingly I am taking the liberty of presenting a dissenting opinion quite frankly and without prejudice.

One of the chief objections to the theories of hearing is that an attempt is being made to visualize how an auditory apparatus might operate and in so doing the physical possibilities have been somewhat over-emphasized and the biological probabilities more or less neglected. May I point out wherein visualization of the available apparatus has seemingly led us astray before applying the actual biological tests of audition to the several theories proposed.

All theories of hearing agree in two major essentials. First; the drum membrane and ossicular chain afford the most efficient route for the transmission of air vibrations to the cochlea for analysis. And second; while the auditory epithelium lies functionally intermediate between the sound stimulus applied and the impulses sent into the central nervous system, this epithelium shows no structural adaptation which may be associated with pitch analysis, let alone any structural differences which may be correlated with discrete frequency response.

The discovery that the ear ossicles form a bent-lever system connecting the drum membrane with the oval window immediately lent itself to the interpretation that the apparatus was more efficient at low frequencies. This harmonized with the findings in the frank conduction deafness of the otologist. However recent investigations appear to show that the bent-lever form of sound transmission system is not

associated with the classical description of decrease in amplitude and increase in force of the drum membrane vibrations in their propogation to the labyrinth. Audiometric methods seem to indicate that the sound transmission apparatus is at least equally efficient at all ordinary audio-frequencies, and if anything perhaps somewhat more efficient at the high end of the sound spectrum. In a word, the entire mechanics of the middle ear apparatus is susceptible to an entirely different functional interpretation than the commonly accepted one. Physically there is little doubt but that the middle ear apparatus overcomes in part at least the enormous impedance differences found in air and in water. Biologically, it is true that acuity of hearing is lowered when the sound transmission system is crippled.

It is a fact that the auditory cells show no structural adaptation which may be correlated with pitch analysis. It is also a fact that the cells show no differences which may be related to the analysis of high frequencies at the base and of low frequencies at the apex of the ribbon-like end organ. The failure to see adaptation however does not rule out the possibility of an intrinsic response which is not visible. We expect to see everything in this day and age. We surely would not examine an eye with a stethescope to determine the errors in refraction. And by the same token a physicist might not make careful examinations of ground sections of a copper conductor to discover why certain currents tend to hug the periphery nor would he smell at a carbon button transmitter to find out whether it was alive or dead.

Physiologists tell us of the great difficulty in the translation of the operation of one sense organ into terms of another sense organ. It is therefore biologically possible for high frequencies to be analyzed at the lower end of the cochlea for the same reason that one tastes sweets on the front of the tongue or why the hair stops short, sometimes too short, at the forehead and gradually drifts out on the back of the neck. The fact that a photographic plate cannot perform the physical impossibility of correcting for the fuzziness of focus does not prove that a living retina cannot and does not do it. It is quite notorious that an eye sees far too well for its optical system and it is possible that the ear hears far too well for its seemingly crude construction. But see something we must and if the adaptation is not a visible intrinsic one, then logically it must be a visible extrinsic factor. Let us look over the adventitious scenery which may therefore explain pitch analysis.

The obvious failure of the Helmholtz basilar membrane conception to meet the requirement of a ten octave analyzing harp has been com-

pensated for in a number of ways. Probably the most satisfactory adjustment is the one presented by Dr. Fletcher. Here a selective weighting factor has been added in the two columns of liquid applied to the basilar membrane and the inter-action of the two windows. High frequencies are therefore analyzed at the lower end of the cochlea and lower frequencies requiring greater weighting at the distal end. I do not doubt that the explanation is quite satisfactory from the standpoint of the physics, and therefore, as a man of much faith I apply biological tests to discover whether or not it actually does explain the facts. If the selective weighing through inter-action of the two windows determines true pitch hearing, then anything which interferes with the interaction of the two windows will produce a false pitch discrimination. Let us see what the experimental evidence may be in the only form available for such study—the living human being. Barany loaded the membrane of the round window with mercury and Oppikofer placed a pellet of wet cotton against it in two cases where the area was exposed. The three cases reported an increase in the acuity of hearing but no change in pitch. Kranz and I had the opportunity of examining a case before and after a bilateral operation which was undertaken to improve the hearing. The drum membrane, and two outer ossicles were removed and the tendon of the muscle attaching to the Stapes was severed. The acuity curve before and after the operation were identical. The woman was naturally depressed on the account of the failure. A pellet of glycerine cotton was applied to the Stapes, which was moveable, and a great enhancement in air acuity obtained although no change in the pitch discrimination. Now how did we know the stapes bone was moveable and how did we know that the cotton prosthesis was applied to it? We know this because the increased acuity occurred only when a certain stop was touched and when this spot was touched the woman became dizzy and showed an eye imbalance which is quite characteristic in sudden changes in labyrinthine pressure. It is very common to find effusion in the middle ear which may mount above the level of the round window and yet Politzer found in cases with unilateral effusion there was no change in the pitch discrimination on the two sides. I have personally done pressure experiments on over three hundred normal and deafened people and the only change reported has been one of lowered acuity. Cases of bony fixation of the stapes are quite common and in spite of the marked interference with the inter-action of the two windows false hearing is exceedingly uncommon. Further in the deafness which comes with age, there is progressive loss of sensitivity beginning

at the high frequency end. This sensitivity loss is accounted for by O. Mayer by a progressive thickening and stiffening of the basilar membrane but such cases do not show false pitch discrimination. The experimental evidence on the living human being appears to indicate that interference with the interaction of the two windows has an effect on the intensity of the sound heard but not on the pitch. Accordingly a biologist would favor the interpretation of this interaction of the two windows as one having to do with intensity even though it would be be necessary to modify Dr. Fletcher's proposal by putting the two windows into full phase rather than half phase relations.

The conception of the action of the basilar membrane as a sound analyzer was originally suggested because the auditory cells showed no visible structural adaptation to their function. Just so the theory advocated by Dr. Shambaugh arose because many investigators felt that the basilar membrane theory could not obtain. It is true that the tectorial membrane is better developed at the distal end of the cochlea but the dimensional differences throughout its length are no greater than those of the basilar membrane. The tectorial membrane is a semi-gelatinous and almost structureless membrane with a specific gravity about that of the liquid in which it is immersed. It therefore appears poorly adapted from a physical standpoint to behave in a manner essentially different from that of the liquid contents of the cochlea. Morphologically, as Dr. Shambaugh has already pointed out, it is closely related to similar structures found in the balancing organ from which the auditory apparatus was derived. This satisfies the biologist on the basis that "things are as they are." Dr. Shambaugh has also mentioned that it may be necessary to abandon an extrinsic resonation theory to which I agree.

We may turn our attention, briefly, to the non-specific activation theories and take for an example the sound pattern theory of Ewald. However if we are to agree to a non-specificity in the auditory cells themselves and a non-specific response in the basilar membrane, then we must complicate the picture by adding a specificity in the nerve impulses which is contrary to all physiology of the nervous system. Nor will the physiologists agree to Dr. Troland's suggestion that coördination of the action currents in the nerve fibers will compensate for the failure of a single nerve fiber to transmit high frequency pulsations demanded in his interpretation. Still further if we are to agree in a cortical analysis of frequency, we must explain away the unique case reported by Dandy where removal of one cerebral hemisphere had practically no effect on the pitch discrimination of the ear of the oppo-

site side. Perhaps it will be well to settle the physiology of the definite physical factors of pitch and intensity before attempting to interpret the subjective additions demanded by the psychologists.

Personally I am experimenting on deafened people in Buffalo in the hope that certain facts may be developed in relation to the intensity of sounds heard. Very few deafened people show any serious disability in so far as pitch discrimination is concerned and a very large percent of deafened people do show evidence of possessing internal ears in which both pitch range and threshold are quite normal.

Theories of hearing have been proposed in many forms. In closing I take the liberty of presenting an explanation of audition in poetic form by Fletcher; not Harvey Fletcher, 1929 but Phineas Fletcher 1633. I am presenting it because few have ever heard of the "Purple Island" and the two short verses are in my opinion excellent.

"As when a stone, troubling the quiet waters,
Prints in the angry stream, a wrinkle round
Which soon another and another scatters
Till all the lake with circles now is crown'd
All so the aire struck with some violence nigh,
Begets a world of circles in the skie;
All which infected move with sounding qualitie.
These at Auditus' palace soon arriving,
Enter the gate and strike the warning drumme;
To these three instruments fit motion giving,
Which every voice discern; then that third room
Sharpens each sound and quick conveys it thence;
Till by the flying poast 'tis hurried hence,
And in an instant brought unto the judging sense."

Canto v 47 and 48

DISCUSSION
SYMPOSIUM ON HEARING

By F. W. Kranz
Riverbank Laboratories, Geneva, Illinois

I believe that we all appreciate the importance of the sense of hearing in our daily life and appreciate the usefulness of the ear in the reception and interpretation of sounds. At least a part of the present discussion is concerned with an appreciation of the ear as a mechanism to serve certain purposes. There are two characteristics of the ear regarded as a physical instrument for sound reception and interpretation which are quite remarkable. One of these characteristics is, of course, the ability to analyze a sound into its component frequencies. The location of the analyzing mechanism or analyzing function has been the subject of much speculation and thought. As one may gather from having listened to the papers just given, most opinions have at least heretofore leaned toward a mechanical method of analysis, this mechanism being located in the inner ear and the results of this analysis being sent to the brain as separate components. In a discussion it is perhaps legitimate to point out in a general way what is involved and also some of the difficulties of this picture. The difficulties are quite apparent to those who have dealt with mechanical or electrical systems having natural periods covering such a wide range of frequencies as is covered by the human ear. This range is approximately from 20 vibrations per second to 20,000 vibrations per second, a matter of ten octaves. In terms of a piano key-board, the human hearing goes about one half an octave lower than the lowest note on a piano and something more than two octaves higher than the highest note on a piano. In a piano, the frequency of vibration of any one string is governed by three factors, the length of the string, the mass of the string per unit length, and the tension of the string. Those familiar with piano construction can readily appreciate the mechanical difficulties which would be encountered in extending the range of the piano by about three octaves. An extension of the range on the low frequency end would call for longer and heavier strings. An extension of the range on the high frequency end requires shorter strings with a higher tension than those now used. An extension of the range by three octaves would be a difficult mechanical problem and when this piano of extended range is imagined to be made in

miniature so that it can be put into a case less than $\frac{1}{2}''$ in diameter, I believe that any mechanic would say that it was impossible. In the case of the basilar membrane of the ear, variations in the natural periods of the component parts are conceived of as being caused in part by the variations in loading or effective mass attached to the vibrating part by that part of the water of the inner ear which is moved by the vibration. The amount of water which acts to load a given part of the basilar membrane is considered as proportional to the path from the oval window to the membrane section and thence to the round window. This would give a greater range than would be possible without the water loading but would still mean that all of the mechanical elements necessary for the complete range of tuned systems were contained within the same small volume before mentioned.

The accuracy of the adjustments, whatever they are and wherever they are, by which various frequencies are differentiated, is indicated by the very small change of pitch which can be readily and definitely detected, the detectable differences amounting to one quarter of one per cent change in pitch at some frequencies. This makes still more difficult the analogy of a piano mechanism and throws a still greater burden upon the mechanical design.

The permanence of adjustment of the frequency differentiation mechanism is another factor of vital importance in considering the probabilities of how the frequency analysis is accomplished. If we had but one ear, it would be rather difficult to show that the absolute adjustment is permanent, for it is possible that the whole scale of tuning could be shifted up or down and we might not be conscious of any change. We would, of course be conscious of any disturbance in the relative tuning of the elements such as might tend to cause an octave or other normal musical interval to be misjudged. However, we have two ears so that the check up of constancy of adjustment is always with us for while it is possible, nevertheless, it is highly improbable that a shift of tuning of the mechanism of the two ears would take place in such a fashion as to keep them alike.

So a mechanism which is to meet the requirements of sound analysis as well as our ears do, must have a frequency range of about ten octaves, must be very accurate in adjustment and must be essentially permanent in adjustment over a period of at least fifty years. This certainly seems rather difficult to achieve in the limited volume of the inner ear and having as the sole materials of construction, the rather soft tissues which are found therein.

Another point which it seems may be commented on in considering the possibilities of mechanical tuned systems is the presence in some people of very abrupt changes in sensitivity, which are formed when the hearing is tested with a continuous tonal range. This abruptness makes difficult the concept of a frequency perception being due to a spread or region of the basilar membrane rather than a very narrowly restricted band and makes perhaps more difficult the idea that pitch perception is due to a vibration pattern of the basilar membrane. Changes in sensitivity are found which amount to changes of as much as 1000 in required energy for hearing with a change in pitch of a semitone.

I noted on one of Dr. Fletcher's slides which gave the possible selectivity which he calculated for the receptor system, that at 2000 cycles, where the selectivity is fairly good, there was a drop of one third of an octave on the low frequency side and two thirds of an octave on the high frequency side for a three bell drop in response, this being a ratio drop of 1000 in energy. This selectivity is not sufficient to account for the experimental facts.

The second characteristic of hearing which is quite remarkable from a purely mechanical standpoint, is its tolerance for a very wide range of intensities. The ratio of intensities between a painfully loud sound and a barely audible sound is approximately a million million to one. This means an amplitude ratio of the square root of this, or a million to one. This would be somewhat equivalent to a scale on which we could weigh a ton of coal and which would also be useable for weighing a mass of a few grams. Attempts have been made to explain this tolerance in terms of variation in nerve impulses sent to the brain from the ear but this explanation still leaves the burden of the million to one amplitude ratio in the inner ear. It is known that the muscles attached to the ossicles of the middle ear exercise a very considerable restraining influence against excessively large vibrations which cuts down considerably the necessary ratio of amplitude for loud and weak sounds. In considering this problem of intensity tolerance with Dr. Pohlman, a further factor of some possible importance has seemed apparent. This is the difference in pressure exerted on the oval and round windows by an incoming sound. For weak sounds, the vibrations are transmitted to the oval window by means of the chain of ossicles without restraint from the ossicular muscles. For very intense sounds, this path of transmission is retarded by means of the ossicular muscles while at the same time the air vibration which is set up in the middle ear by the vibrations of the drum membrane impress themselves on the round window so that

the round and oval windows are both affected by the incoming sound. These opposing forces tend to keep down the motion of the fluid in the inner ear and so also the amplitude of the basilar membrane. I do not know whether or not this explanation has appeared in the literature but it seems that it is worthy of consideration.

It was perhaps due to lack of time that Dr. Fletcher did not explain the justification for cutting off about an octave and a half on each the high and low frequency ends of the frequency range in his consideration of frequency response. He considered a frequency range from 88 cycles to 8000 cycles instead of the known frequency range of 20 cycles to 20,000 cycles.

Dec. 13, 1929.

THE HUMAN ELEMENT IN PIANO TONE PRODUCTION

By William Braid White
American Steel & Wire Co.

For some time past the Acoustic Laboratory of the American Steel & Wire Company has been conducting an investigation into the behavior of steel wire under tension, with special reference to the particular case of steel wire used for musical purposes, such as piano wire. In the course of the investigation it became necessary to obtain records of the sounds produced by pianists at the piano. To this end a Westinghouse portable single-element Osiso was obtained, and after various experiments was made available for the special purposes of the investigation. The particular piece of work described in this paper consisted of obtaining various records of sounds produced by different musicians under different prescribed conditions, but always as nearly as possible as those sounds would normally have been produced in the course of piano playing. The cooperation of numerous eminent artists has been obtained. One special phase of the investigation has been devoted to inquiring into the relation between intensity and quality of sound. Many hypotheses, some of them obviously incorrect, have been held by teachers of piano as to the possibility of producing, by means of variations of what is called "touch," sounds of different qualities or colors, irrespective of the intensity or loudness-value of each such sound. The object of the special investigation here described was to ascertain some of the facts in the case.

Study of the mechanism or "action" of the piano discloses the fact, known to all technical workers on it, but apparently not known to all musicians, that its strings are excited into motion by blows from pivoted "hammers," of which the duration is always only a very small fraction of a second. Each such blow is actually and necessarily always delivered before the finger of the musician has pressed the corresponding key down to the bottom of its $\frac{3}{8}$ inch travel. Thus, by the time the key has reached the end of its travel, the hammer which has struck the blow has already rebounded from the string and has been caught by the checking devices, to be held thus until the key has been released for another stroke. It follows that the musician can exert no control over a sound produced by him through the key and "action," at any time after the immediately-rebounding hammer stroke has been

delivered. During that portion of his manipulation of the key which precedes the stroke of the hammer, the musician can depress the key (a) with constant velocity, (b) with velocity increasing as the key comes down or (c) with velocity decreasing during the downward travel. It will be remembered that the total distance through which the finger-end of the key travels as it actuates the hammer is $\frac{3}{8}$ inch, and that the distance traveled up to the moment of the production of the sound is even less than this, the actual distance depending upon the mechanical adjustment of the action.

It is evident, then, that the musician can cause the hammer to travel towards the string at any one of many possible velocities, and that the velocity of the hammer at the moment of its contact with the string is the one factor in production of tone which the musician can control. This statement, of course, is to be taken as referring strictly to single acts of key-depression, and to each separate individual sound taken by itself. It is well known, of course, that the use of the sustaining pedal introduces additional physical elements which affect the sensations experienced by the player and the listeners. It is also well known that the position of any individual musical sound in a group of such sounds forming part of a musical sound-sequence, as well as its duration in relation to the durations of adjacent sounds, powerfully affects the impression given to the hearers. These elements, however, are secondary and must not be considered in the present discussion.

When the piano string is exicted into vibration by the blow of a piano hammer, then, immediately after its whole-length vibration has developed, it invariably subdivides into segmental vibrations, which produce sounds at frequencies related to each other as the terms of the series 1.2.3.4.5. . . . . n. The quality or color of sound given out by a piano string, or by any other source of musical sound, is well known to vary with the number and relative intensity of the partial tones thus set up. In the case of a piano string, its length, diameter, unit-weight tension and the point at which the stroke is made are all fixed. Thus, it may be said that a blow which should give a maximum displacement of the string would evoke from it all the possible partial tones which it might be capable of emitting according to the physical factors just now mentioned. Evidently, therefore, all types of displacement produced by blows, from a maximum down to zero, must represent a decreasing series of groups of partial tones, for obviously the smaller the displacement, the fewer partial tones will rise above the

threshold of audibility. Hence, the fainter the sound, the simpler its composition, and the nearer to a pure fundamental sound it becomes. On the other hand, the louder the sound, the more partial tones become audible, and consequently the more complicated is the form of vibration and the more highly organized the quality or color of the sound.

Now since the amplitude of a string's vibration measures the loudness, and since also the actual physical shape of the sound wave represents from moment to moment its combination of fundamental and partial tones, (that is, its color), it is evident that, if records can be made which will show visibly these changes in amplitude and in shape, the resulting photographs thereof can be analyzed and interpreted. This has been done in the course of the present investigation, and some of the photographs obtained are here shown.

Fig. 1 represents a fundamental form of vibration, drawn from the motions of a large tuning fork giving 440 vibrations per second. It is included here mainly as a reminder that all musical sounds, when

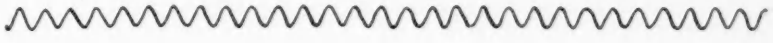


Fig. 1.—Simple sine wave-form from tuning fork.

analyzed into their component parts, are found to consist of a greater or smaller number of perfectly simple sounds of this type, each such component sound being the result of the simple harmonic motion of the corresponding segment of the string, air-column or whatever is the oscillating body.

Fig. 2, a, b, and c, represents three evocations of sound from the same key of a concert grand piano, by Mr. John Powell. The key is B, No. 39 counting upwards from the bass end, one semitone below middle C. The pitch is 246 v.p.s. In each case the key was depressed by the third finger of the artist's right hand. Mr. Powell described his physical act as "touch delivered with high, rigid wrist." He also described, before making it, the tone quality sought by him in each case, as "Hard tone." The first sound (a) was given at high intensity, or in musical language "forte." The next (b) was done at "mezzoforte" intensity and the third (c) at low intensity or "piano."

Two points stand out upon examination of the pictures. In the first place the gradations of sound intensity were perfectly apparent to the

listeners. In the second place, these differences of intensity are registered on the pictures as differences of amplitude. In the third place each picture shows a markedly individual shape.

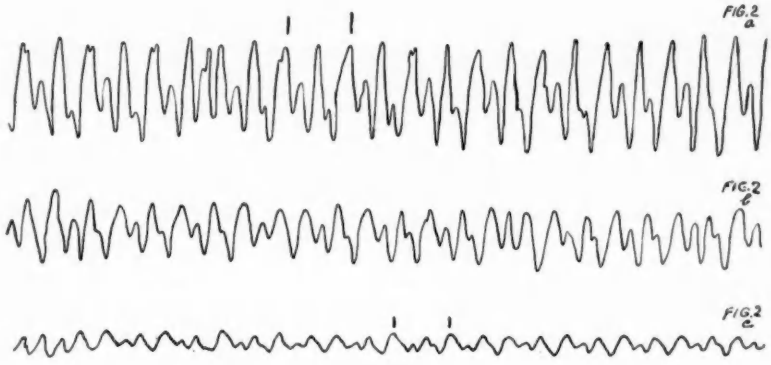


Fig. 2.—Variations in tone production at piano by John Powell.

The three curves were submitted to the Riverbank—Laboratories of Geneva, Illinois, for analysis on their Henrici apparatus. The work was carried out to 15 components in each case by Mr. B. E. Eisenhour. The table below gives the amplitudes in millimetres of each component from #1 to #15, as analyzed out from one complete cycle of each sound-curve, after enlargement in each case to the scale of the machine, which is 400 mm per wave-length.

Component No.	a (loud)	b (medium)	c (faint)
1	68.4	34.7	19.2
2	186.0	102.0	42.5
3	90.8	57.2	21.1
4	91.0	53.0	9.8
5	54.7	44.4	11.4
6	41.4	7.5	1.0
7	16.3	3.6	1.0
8	31.0	4.9	0.9
9	4.1	4.4	0.6
10	14.8	5.1	0.6
11	1.7	2.8	0.4
12	4.1	2.1	0.8
13	5.9	1.4	0.2
14	4.4	0.8	0.1
15	5.7	2.6	0.3

In order to render more vivid the difference of "quality" developed in each case as a direct consequence of the three different amplitude values, the following diagram is given. (Fig. 3.) Here the ordinates represent the comparative amplitudes of the components and the abscissas their succession from #1 to #15. By joining the extremities of the ordinates we obtain a diagrammatic suggestion of the three different sound-forms in the shape of three markedly individual curves. Other analyses of the kind have been made, which confirm the expectation that differences of curve-shape correspond to differences of component-structure, with corresponding differences of tone-color or quality. It is of course well known that difference of phase will produce

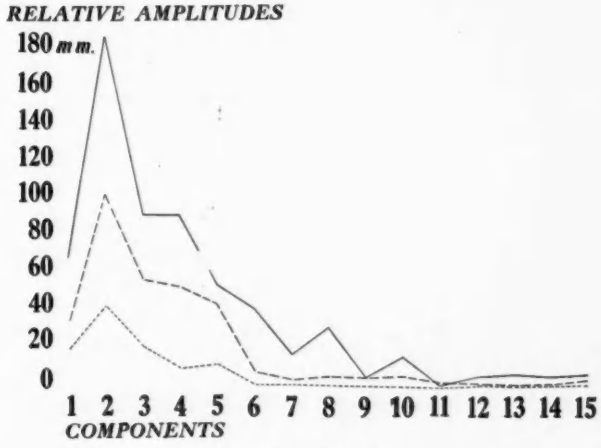


Fig. 3.

difference of curve-shape without reference to any other difference of component-structure, but in the case of the piano, where all factors are constant save only hammer-velocity (i.e. for any single isolated sound), phase differences do not enter in.

We are thus left with evidence in support of the hypothesis that every difference of hammer-velocity connotes a difference both of intensity and of tone-color. Since "touch," as we know, is a matter of hammer-velocity-control, it follows that every separate process of muscular action on a key which produces a corresponding individual hammer-velocity, produces consequently both an individual loudness-value and an individual component-structure-value, (in musical language an individual color value). To put it in another way we may say that:

(a) "touch" on the piano is control of hammer velocity.

(b) Control of hammer-velocity is, primarily and fundamentally, control of loudness, or amplitude of string-motion. At the same time:

(c) every individual amplitude value results in a wave form of individual shape, so that:

(d) every loudness-value has a corresponding distinct individual color-value.

As a natural corollary it may be said that:

(a) there is no change of loudness without corresponding change of color:

and it likewise follows that:

(b) there is no change of color without change of loudness.

On this view, it is not possible for a pianist to obtain from the same note two sounds of different color-values (component-structures) without a corresponding change of loudness. On this point additional testimony is available. Fig. 4 shows an experiment conducted by the aid of Mr. Rudolph Ganz. This artist was asked to make three sounds,

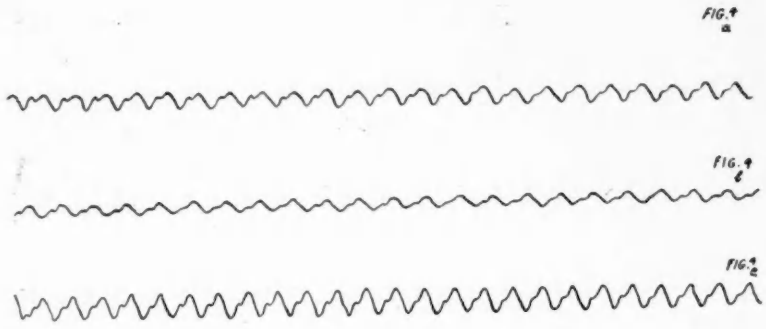


Fig. 4.—Three tones by Rudolph Ganz made at the piano.

using the same key of the same piano, (No. 39, first space above bass clef) as nearly as possible alike in loudness, and if possible of identical loudness, but to impart to each a tone-color which to his ear should be obviously different in each case. Mr. Ganz kindly made the attempt after taking all due precautions to assure himself that he actually was perceiving differences of color. The result came out as shown in Fig. 4.

It is evident that the three differences of shape are present, corresponding to differences of tone-color. It is equally obvious that the three amplitudes are not identical, showing that the loudnesses are not the same.

Similar experiments made again with the aid of Mr. John Powell have given corresponding results even more definite, and have thus tended to confirm the hypothesis as to the dependence of tone-color differences upon corresponding differences of loudness.

These few pictures are merely a small selection among many dozens which have been taken. During the last few months more than fifty separate experiments have been undertaken with the cooperation of Harold Bauer, Joseph Lhevinne, Rudolph Ganz, Vladimir Horowitz, Mischa Levitski, John Powell, Ernest Schelling, Rudolf Gruen, Abram

Fig. 5.—Rudolph Ganz recording his tone production.

Chasins, Carolyn Beebe, Olga Samaroff, Sigismond Stojowski, Alexander Siloti and other artists, for the purpose of developing the evidence. Among other things it may be mentioned that Mr. Levitski succeeded in getting at one key of a piano, 21 attempted, and 18 actual, differences of loudness, from the faintest to the loudest he could obtain. Each shows its own individual amplitude and each its own individual curve-shape: i.e. each has its own loudness and its own color.

Fig. 5 shows a typical set-up for taking these records. The musician (Mr. Rudolph Ganz) is seen seated at the piano. Across the keyboard

is placed a frame containing a moveable contact which can be adjusted so as to flash on the photographic lamp of the Osiso at any point during the descent of the key. The microphone used is a Western Electric 387-W. The current from the microphone is amplified by passage through a set containing two 171 and one 112-A vacuum tubes (D.C.). The photographic lamp consumes 2 amperes at 8 volts.

These investigations have thrown, and are throwing, a valuable light upon the art of piano playing. It is evident that the musician thinks in terms of his mental and physical acts, and he naturally tends to frame verbal hypotheses to explain the effects he obtains, hypotheses which describe his own feelings and therefore are commonly vague and sometimes even contradictory. It is necessary to keep in mind clearly the distinction between the two points of view. It is perfectly correct to say, for instance, that what the pianist may describe as a wrist movement, as opposed to an arm or finger movement directed at the key, may have virtues of its own as a producer of tone from the piano; but it is necessary to remember that this is simply another way of saying that he has learned how to impart various velocities to the hammers of the piano. He does not think in terms of hammer velocity, but of his mental processes and physical movements. He may find it hard to believe that all these reduce themselves to so many elaborate devices for controlling hammer velocity, that is to say loudness of sound; yet it should be a sufficient consolation to him to be able to say, as he may, that each difference of loudness involves a difference, however slight yet distinct, of tone-color. Thus the operation of producing tone from the piano aims, for its primary objective, at control of one factor only, namely hammer-velocity, or in other words of string amplitude; or again in other words, of loudness. Such control nevertheless may be extremely delicate and capable of many gradations (videlicet the Levitski experiment), while there is the assurance that as many varieties of tone color may be obtained as the pianist can obtain varieties of loudness.

In point of fact it appears that a similar correlation obtains throughout the range of musical instruments; but that is something to be demonstrated at a later time.

These experiments are continuing and results will be announced from time to time. It is our opinion that the reduction of the process of controlling the tone from the piano to the process of controlling hammer-velocity does not in any way detract from the beauty of the art, since it shows, among other things, what extreme delicacy of control

is called for, and, in turn, to what an extent a great artist is able to bring his command over his mental and physical processes to bear upon the task of obtaining almost infinitesimal varieties of manipulation of a key-board, no one of the 88 members of which can travel through a greater distance than $\frac{3}{8}$ inch.

Note: The curves shown in Fig. 3 have been drawn from the amplitude only of the harmonic components of the sounds. If the intensities had been plotted instead, on the formula $I = (AF^2)$ where I is the intensity, A the amplitude and F the frequency of each component, the curves of Fig. 3 would show different shapes, but the argument would not thereby be vitiated.

AN INSTRUMENTAL METHOD OF REVERBERATION MEASUREMENT

By R. F. Norris, *C. F. Burgess Laboratories, Inc.*, and C. A. Andree, *University of Wisconsin*

If research work is to be carried on in any art, the first concrete need the worker realizes is that of definite units of measurement. In the study of auditorium acoustics Wallace Sabine has given us a standard of inestimable value in his period of reverberation. He has defined the period of reverberation of a room as the time in seconds it takes a sound of any average initial intensity to die away to one millionth of that intensity.

Of what value is this unit in the study of auditorium acoustics? Its chief value is obviously that it may be applied to all auditoria, irrespective of size, shape and finish. That is, it is a property that is equally germain to all spaces enclosed in each of the three dimensions, however large or small they may be and regardless of the material nature of their walls.

A large number of auditoria of various sizes and shapes, which are known to be good acoustically, have been measured by this unit. From these measurements, it has been determined that in large rooms a longer period of reverberation produces good audition, while in smaller rooms a shorter period is desirable. Since the period of reverberation of a room is the yard stick by which its acoustic properties are measured, it becomes of great importance to the acoustical engineer in his work. It is an undoubted fact that the more information obtained concerning this property of all sorts and sizes of rooms, when linked with their observed acoustical properties, will produce a wealth of valuable information on which a more exact understanding of the subject of acoustics may be founded.

If we are to base our knowledge of auditorium acoustics on the study of periods of reverberation, it is evident that we must have a means of determining these periods. Therefore, a discussion of the method usually employed is not amiss at this time. The extreme simplicity of this method is its highest recommendation. The procedure briefly is this: A calibrated organ pipe is caused to speak in the room for a time long enough to enable the average sound to build up to a constant value. The pipe is stopped suddenly and a stop watch is started simultaneously. The observer listens carefully and stops the watch when the sound

has finally died out to complete inaudibility. This routine is repeated several times and from an average of the readings so obtained the period of reverberation is computed. Corrections must be made for the calibration of the pipe and the absorption of the room. The chief disadvantages of this method are that we must assume the constant sensitivity of the observer's ear, and that small extraneous sounds often so mask the decaying sound that the readings may be in considerable error. Then, again, the period of reverberation of the room must be of an appreciable duration or the reading cannot be made, and it is very often impossible to obtain absolute quiet in a room to be measured.

After employing this method for some time, the Burgess Laboratories, realizing its limitations, endeavored to discover an instrumental method which would eliminate the human element, and upon which small, extraneous sounds would have no effect. As a means of measuring sounds, an instrument known as the acoustimeter was built. This instrument comprises a magnetophone type of pick-up, an audio amplifier and a very sensitive thermal voltmeter. As a sound source, first an electric tuning fork, and finally an audio frequency oscillator feeding into a loud speaker was used. The proposed procedure was to start the source and with the acoustimeter measure the level to which the sound built up. Experiments showed that while the theory was correct that sound patterns were set up in the room to such an extent that any desired reading could be obtained by the judicious placing of the pick-up. Attempts were made to destroy these patterns by rotating the loud spearker used as a source and by continuously varying the frequency of the emitted sound over a range of 25 cycles. This helped in a great measure, but the results were still anything but satisfactory. It was found, however that if the source was sounded continuously, that sound intensities could be charted throughout the room under study, and that the resulting data were valuable in studying the effects of the shape of the room on the sound density pattern.

A further analysis of the problem lead to a new procedure, by which the period of reverberation of a room may be accurately and easily measured. This procedure is as follows: A source is sounded for a definite length of time, and at exactly uniform intervals. During an exact portion of each of the intervals of quiet, the sound measuring device is turned on, and the average residual sound is measured over that portion. Thus a reading is obtained which is proportional to the area under the sound decay curve, over a predetermined time. The time of measurement is now shifted bodily to a different portion of the decay curve,

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and a second reading is taken, which is proportional to the area under the sound decay curve over the same length of time but under a different portion of the curve. If the first reading is taken during an interval of time which begins very shortly after the sound is stopped, and the second reading is taken during an interval which is t^1 seconds later, it is evident that the first reading will be larger than the second. To determine the significance of this difference the ascending and descending sound curves must be studied.

It is a well known fact that when a steady sound is emitted in a room, the energy in the room builds up to a definite, average level determined by the power of the source, and the properties of the room. This building up of sound follows the exponential formula; $E=E_0(1-e^{-avst/4V})$, in which E is the instantaneous value of the average energy, E_0 is the maximum value which E approaches, e is the naperian base, a is the Sabine coefficient of absorption, v is the velocity of sound, s is the exposed surface of the room, t is the time in seconds, and V is the volume of the room. When the sound is stopped, the average intensity of the room dies off in a like manner, the equation being $-E=E_0 e^{-avst/4V}$.

In the proposed procedure, when the sound is started for the first interval, the average intensity in the room will rise exponentially to a value of E^1, which may be less than the maximum E_0. The sound is then stopped, and the average intensity decreases in like manner to a value of E_1. The sound is then turned on again, and the average intensity rises to a slightly greater value than E^1 due to the addition of some of the residual sound from the first cycle.

If the time interval between the stopping of the sound, and the start of the sound measurement interval, be designated as t_3, and the time interval between the stopping of the sound and the end of the sound measurement interval, be designated as t_2, and the time of the total cycle be T, then the average sound measured during n cycles may be expressed as

$$E_{ave}=\frac{1}{nT}\left\{\int_{t_3}^{t_2} nE_0'e^{-avst/4V}dt+\int_{T+t_3}^{T+t_2}(n-1)E_0'e^{-avst/4V}dt\right.$$
$$+\int_{2T+t_3}^{2T+t_2}(n-2)E_0'e^{-avst/4V}dt\cdot\cdot\cdot+\int_{(n-2)T+t_3}^{(n-2)T+t_2}2E_0'e^{-avst/4V}dt$$
$$\left.+\int_{(n-1)T+t_3}^{(n-1)T+t_2}E_0'e^{-avst/4V}dt\right\}. \qquad (1)$$

Let $K=avs/4V$ and then let $E_0'=E_0(1-e^{-kt_1})$ then (1) becomes

$$E_{ave}=\frac{1}{nT}\Big\{\int_{t_3}^{t_2} nE_0'e^{-Kt}dt+\int_{T+t_3}^{T+t_2}(n-1)E_0'e^{-Kt}dt$$
$$+\int_{2T+t_3}^{2T+t_2}(n-2)E_0'e^{-Kt}dt\cdots+\int_{(n-3)T+t_3}^{(n-2)T+t_2}2E_0'e^{-Kt}dt$$
$$+\int_{(n-1)T+t_3}^{(n-1)T+t_2}E_0'e^{-Kt}dt\Big\}. \qquad (2)$$

$$E_{ave}=\frac{-E_0'}{KnT}\{ne^{-Kt_2}-ne^{-Kt_3}+(n-1)e^{-K(T+t_2)}-(n-1)e^{-K(T+t_3)}$$
$$\cdots+2e^{-K[(n-2)T+t_2]}-2e^{-K[(n-2)T+t_3]}+e^{-K[(n-1)T+t_2]}-e^{-K[(n-1)T+t_3]}\} \qquad (3)$$

$$E_{ave}=\frac{E_0'(e^{-Kt_3}-e^{-Kt_2})}{KnT}\{n+(n-1)e^{-KT}+(n-2)e^{-2KT}$$
$$\cdots+2e^{-K(n-2)T}+e^{-K(n-1)T}\} \qquad (4)$$

Let the bracketed expression be designated as y

$$y=1+e^{-KT}+e^{-2KT}+e^{-3KT}\cdots+e^{-(n-1)KT}$$
$$1+e^{-KT}+e^{-2KT}+e^{-3KT}\cdots+e^{-(n-2)KT}$$
$$\cdots\cdots\cdots\cdots$$
$$\cdots\cdots\cdots$$
$$\cdots\cdots$$
$$1+e^{-KT}$$
$$1$$

Writing the sum of these geometric equations

$$y=\frac{1-e^{-nKT}}{1-e^{-KT}}+\frac{1-e^{-(n-1)KT}}{1-e^{-KT}}+\cdots+\frac{1-e^{-2KT}}{1-e^{-KT}}+\frac{1-e^{-KT}}{1-e^{-KT}}$$
$$(1-e^{-KT})y=n-e^{-KT}\{e^{-(n-1)KT}+e^{-(n-2)KT}\cdots+e^{-2KT}+e^{-KT}+1\}$$
$$(1-e^{-KT})y=n-e^{-KT}\frac{(1-e^{-nKT})}{(1-e^{-KT})}$$
$$y=\frac{n}{1-e^{-KT}}-e^{-KT}\frac{(1-e^{-nKT})}{(1-e^{-KT})^2}.$$

Putting this value for y in our original expression number (4)

$$E_{ave}=E_0'\frac{(e^{-Kt_3}-e^{-Kt_2})}{KT}\left[\frac{1}{1-e^{-KT}}-\frac{e^{-KT}(1-e^{-nKT})}{n(1-e^{-KT})^2}\right] \qquad (5)$$

Let $n=\infty$. Then the second term in the square brackets approaches 0 and we have

$$E_{ave}=\frac{E_0'(e^{-Kt_3}-e^{-Kt_2})}{KT(1-e^{-KT})} \tag{6}$$

Substituting the value of E_0' in terms of E_0

$$E_{ave}=\frac{E_0(1-e^{-Kt_1})(e^{-Kt_3}-e^{-Kt_2})}{KT(1-e^{-KT})} \tag{7}$$

Adding a time interval t' to both t_3 and t_2

$$E_{ave}=\frac{E_0(1-e^{-Kt_1})(e^{-K(t_3+t')}-e^{-K(t_2+t')}}{KT(1-e^{-Kt})} \tag{8}$$

$$E_{ave}'=\frac{E_0(1-e^{-Kt_1})(e^{-Kt_3}-e^{-Kt_2})(e^{-Kt'})}{KT(1-e^{-Kt})} \tag{9}$$

Dividing equation (7) by equation (9)

$$\frac{E_{ave}}{E_{ave}'}=\frac{1}{e^{-Kt'}} \text{ or } e^{Kt'}$$

$$\therefore K=\frac{1}{t'}\log_e\frac{(E_{ave})}{(E_{ave}')}$$

By definition the period of reverberation t_p may be expressed as $E_0/E=e^{-ktp}=e^{-13.8}$. Therefore $Kt_p=13.8$ or $t_p=13.8/K$. Substituting the calculated value for K from the equation above, we have $t_p=13.8t'/\log e\ (Eave.)/E'ave)$ or changing to the common logarithm $t_p=6t'/\log_{10}(Eave)/(E'ave)$. Since the two readings taken are proportional to E'*ave* and *Eave* respectively and the time t' is known, the period of reverberation t_p is determined.

The advantages of this method are apparent. Briefly they are:

(1) Each reading consists of the statistical average of a large number of individual readings, 100 readings being taken per minute.

(2) Each individual reading corresponds to the average area under a considerable portion of the decay curve, which eliminates errors introduced by periodic variations in this curve.

(3) The method is not dependent on the observer's acuity of hearing, nor is it dependent on the sensitivity of the instrument.

(4) Since the sound is decaying while the measurements are being taken, the sound pattern is constantly changing and breaking down, and thus truer averages of the average intensity are obtained.

(5) The intermittent character of the sound very closely approaches the character of speech sounds.

It may be pointed out here that this procedure lends itself admirably to the determination of the acoustical qualities of materials. The absorption of a room may be measured in this manner, both before and after definite quantities of material to be studied have been introduced, a simple subtraction giving the absorption of the added material. If this method is used for the measuring of acoustical properties, a reverberation room of very small dimensions may be employed, since the periods of reverberation can be measured accurately, even though they are too

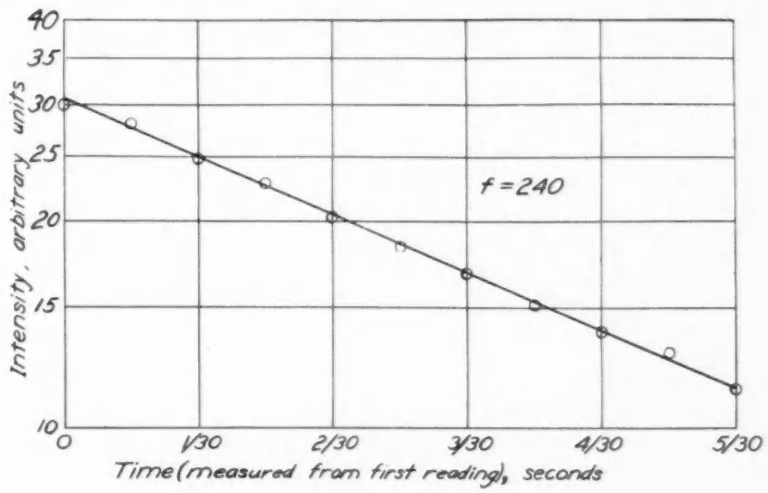

FIGURE 1

short to be measured by the ear. In fact, the method has been used successfully in a room as small as 5′x6′x9′.

The accompanying curves represent two sets of data taken on an empty reverberation room. This room is 5′ x 6′ x 9′ in size. The walls are painted concrete. There are 8 square feet of painted wood work exposed to the sound in the sample supports and frames. A mixing fan of galvanized iron having a total area of 27 square feet revolves at 60 revolutions per minute. In taking the data plotted readings were taken varying t' from zero to 1/6 seconds by 1/60 second steps, five readings being taken at each step. By this procedure points are obtained which serve to check the accuracy of the first and last point from which the period of reverberation is calculated. The two periods as computed from the data are: at 240 cycles per second

$$\frac{6\times\frac{10}{60}}{\log_{10}\frac{6.1}{2.28}}=2.38 \text{ seconds}$$

and at 1400 cycles per second

$$\frac{6\times\frac{10}{60}}{\log_{10}\frac{5.33}{1.26}}=1.597 \text{ seconds.}$$

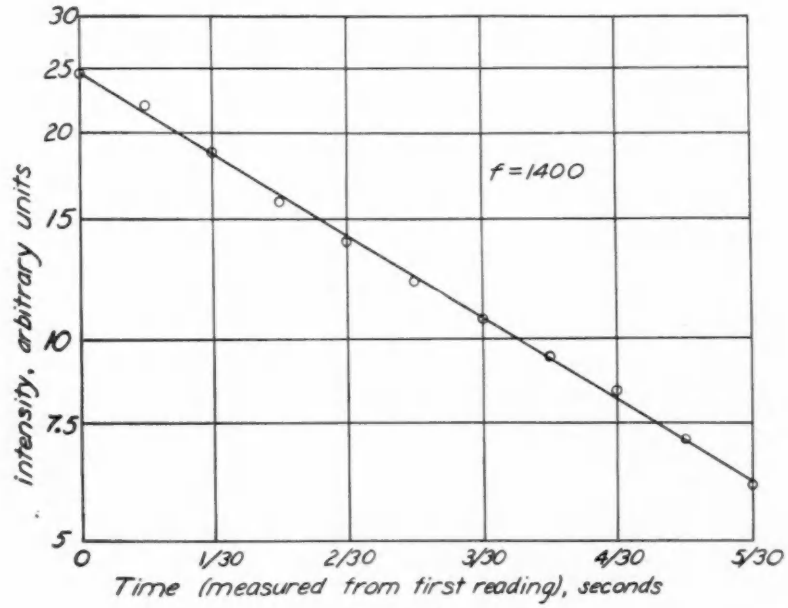

FIGURE 2

It may be pointed out that it would be quite impossible to measure, by the organ pipe method with any degree of accuracy, the period of reverberation of a chamber having so small a volume.

THE STRIKE NOTE OF BELLS

By Arthur Taber Jones
Smith College

Introduction

A bell, like most sounding bodies, usually gives out several different tones at the same time. For the production of each of these tones there is a definite "normal mode" of vibration. The different normal modes are characterized by different numbers of vibrating segments separated by nodal lines. Some nodal lines run up and down the bell, and some are circles parallel to the plane of the mouth.

When a bell is struck its vibration consists of a number of normal modes all occurring at the same time, and in the sound from the bell there are the various corresponding tones. The one of these tones which has the lowest pitch will be called the *first partial* or *first partial tone*, the one of next higher pitch the *second partial*, and so on.[1]

The partial tones of a bell do not in general follow the harmonic series. For instance, on the largest bell of the Dorothea Carlile Chime at Smith College the first seven partial tones have frequencies which are approximately in the ratios 1.00:1.65:2.10:3.00:3.54:4.97:5.33. Thus the partial tones are quite different from those of strings or pipes.

There are a number of ways in which the separate partial tones may be picked up. If a bell is struck and a Helmholtz resonator is held to the observer's ear, it is often possible to pick out some of the partial tones. The resonator may be tuned by rolling a finger over its mouth, and when the natural pitch of the resonator comes near to one of the partial tones of the bell the loudness of that partial increases. A second

[1] It frequently happens that there are two or more normal modes with the same number of vibrating segments. For instance, each of two modes may have four nodal lines running up and down the bell, the nodal lines for one mode lying between those for the other. The stiffness and inertia coefficients for the two modes are likely to differ in such a way as to give rise to two tones of nearly the same pitch. This is the cause of the beats that are often heard from a bell. When two or more modes have the same number of vibrating segments, similarly distributed, and give rise to frequencies so close together as to beat, I shall count them as a single partial tone.

If we were to use optical terms the different modes in a single partial might be regarded as giving rise to "multiple lines." On the bells of the Dorothea Carlile Chime the lower partials are often double, the sixth is usually a multiplet of higher order, and the seventh is sometimes a doublet. In the experimental work described in this article the frequencies of the component "lines" have usually been determined, but the values used are averages.

method makes use of the beats between one of the partial tones and some other source of sound. Suppose, for instance, that a tuning fork has a pitch which differs slightly from that of one of the partial tones of a bell. If bell and fork are struck the beats may be very distinct. A third method makes use of resonance from the bell. Suppose as before that the pitch of a tuning fork is close to that of one of the partial tones of a bell. This time the bell is not struck. The fork is struck and then its stem is pressed against the bell. The bell picks up the vibration and sings out that one tone.

Some of the partial tones of a bell are easily heard without any resonator or tuning fork. They may, in fact, be heard so easily that when a single bell is ringing it is not always easy to decide what the pitch of the bell is. But when several bells follow each other in fairly rapid succession there is usually no difficulty in deciding on the pitches. The note which is most prominent when several bells sound in succession is called the *strike note* of the bell.

More than forty years ago Lord Rayleigh[2] was surprised and disturbed at finding that the strike notes of a number of bells did not agree with the pitches of any of the first few partial tones. For instance, the second bell (reckoned from the highest) of the Terling peal was supposed to be b_3, i.e. the first b above middle c. The first five partials were: d_3-6, $a_3\sharp-5$, d_4+8, $g_4\sharp+10$, and b_4+2, where the numbers added and subtracted indicate the numbers of beats per second made with notes on a harmonium. It will be seen that the only partial which is a b is the fifth, and that this fifth partial is an octave higher than the strike note. The strike note is not one of the partial tones of this bell.

Rayleigh examined the first five partials of seven bells. On all of them the fifth partial was an octave higher than the strike note. On two of the bells the second partial had nearly the pitch of the strike note, but on the others no partial was very close to the strike note.

Ten years ago I published[3] a study of the twelve bells of the Dorothea Carlile Chime at Smith College, and more recently[4] a study of the ten bells of the Harkness Memorial Chime at Yale University. On all of these bells the fifth partial is just about an octave above the strike note. On the bells of the Harkness Chime the second partial is close to the strike note, but on most of the bells of the Dorothea Carlile Chime

[2] Rayleigh, Phil. Mag., **29,** 1 (1890).
[3] A. T. Jones, Phys. Rev., **16,** 247 (1920).
[4] A. T. Jones, Phys. Rev., **31,** 1092 (1928).

the second partial is lower than the strike note and the third partial is higher.

During the study of the Dorothea Carlile Chime it seemed to me that the fifth partial came out very promptly when a bell was struck, and that the second and third partials grew to their maxima more slowly. I suggested that the pitch of the strike note may be determined, except for octave, by the fifth partial, and that the octave in which we hear it may be determined by the more sluggishly responding second and third partials. The work on the Harkness Chime was not decisive, but it seemed to indicate that the fifth partial really does come out promptly, and the third partial not so promptly.

The work last summer was again on the Dorothea Carlile Chime, and was done in the hope of getting further evidence as to the cause of the strike note.

Determinations of Pitch

Frequencies of Partials.—For the work last summer a set of nine tuning forks with adjustable loads was made and calibrated by the Riverbank Laboratories. These forks run from 168 cycles/sec to 1400 cycles/sec. They are graduated so that from 168 to 960 the successive settings differ by four cycles/sec, and from 960 to 1400 they differ by eight cycles/sec. These forks cover the range of the first seven partials on the six largest bells of the Dorothea Carlile Chime, and the work now reported is confined to those six bells.

With these tuning forks the pitches of the first seven partial tones were determined by the frequencies of the beats. On the average the frequencies thus determined differ from those found ten years ago by about one per cent. The present values are much the more reliable. The average deviation of several determinations is now almost always less than a tenth of one per cent.

Frequencies of Strike Notes.—Since a strike note does not beat with a tuning fork, its frequency must be found in some other way. A fork tuned to the octave below the fifth partial of a bell sounds to me a little sharper than the strike note, and it seemed desirable to have the judgments of a number of musicians as to the exact pitch of the strike note. Seven persons with some musical training—several of them professional musicians—were good enough to assist me. Each of them gave me from half an hour to an hour of his time. I selected three bells which are readily accessible, and each of the seven musicians gave me his judgment on each of the three bells. I would adjust a fork to a

pitch near the strike note and then ask whether the fork was sharper or flatter than the bell. This was repeated until it seemed that sufficient data had been obtained.

The results are given in Table I. It will be seen that the average frequency is in each case close to an octave below the fifth partial of the bell. It is also true however that the range from the lowest value on any

TABLE I.

FREQUENCIES OF STRIKE NOTES*

Observer	*f*	*g*	*a*
1	329	380	429
2	340	384	423
3	346	383	434
4	343	387	442
5	346	390	437
6	348	389	439
7	349	392	448
Average	343	386	436
Octave below fifth partial	345	387	438

* The observers are not listed in the order in which they gave their judgments, but beginning with the lowest values and going up to the highest.

one bell to the highest value on that same bell is in each case just about a half step.[5] Several of the musicians said that when fork and bell had nearly the same pitch some difference in the quality of the sounds made it difficult to decide which was the sharper. The average deviation of the values for one bell is in each case a little over one per cent, so that the difference between the average and the octave below the fifth partial is probably without significance. Although the range covered by the separate values is considerable, it may be reasonable to conclude that on these three bells the strike note is fairly close to an octave below the fifth partial.

Griesbacher's Method.—Griesbacher[6] has found that the strike note of a bell can be elicited by a tuning fork of the same pitch. The fork is struck and then pressed against the bell in such a way as to have a

[5] I find that some persons are not familiar with the terms *step* and *half step*. The same intervals are also called *tone* and *semitone*. Since the terms "step" and "half step" are in use I prefer not to use the word "tone" to indicate an interval between two tones.

[6] P. Griesbacher. Glockenmusik, pp. 53–55, Alfred Coppenrath (1927).

very small area of contact. Griesbacher finds it desirable to bring the stem of the fork, or even better a sharp edge near the bottom of one of the prongs, against the lip of the bell—where the inner and outer surfaces meet. I find it satisfactory to hold the axis of the fork at an angle of about 45° with the surface of the bell and then touch the stem lightly to the soundbow—on which the clapper strikes. The particular azimuth at which the fork is held seems not to be important. The tone elicited from the bell in this way is rather faint and of short duration, but the effect is clearly a real one.

Although a fork tuned to the octave below the fifth partial sounds to me a little sharper than the strike note, I find that a fork thus tuned brings out the strike note, and that a fork tuned so that it sounds to me to have more nearly the pitch of the strike note does not bring it out so well. This result is further evidence that the strike note is an octave below the fifth partial.

Beats from Octave of Fork.—It is possible to get beats from a bell and a fork when the fork is struck and the bell is not. Let a fork be tuned so that its pitch is near to that of one of the partial tones of a bell. Let the fork be struck, its stem pressed against the bell, and the fork then removed. Both bell and fork will now vibrate freely, and the tones from the two will beat. Even if the fork remains in contact with the bell beats may often be heard.

When a fork is tuned so as to elicit the strike note there are often similar beats. They are evanescent, but a few may be heard. If different settings of the fork are made and the beats in each case are timed, the frequencies calculated from the beats differ. But if we assume that the vibration of the fork contains the octave,[7] and that this octave beats with the fifth partial, I find that the frequencies calculated from the different settings of the fork are in agreement.[8]

I have used this same method to determine the frequencies of the seventh partials on five bells. This was possible because in the bells of the Dorothea Carlile Chime there is no partial close to an octave below the seventh partial. When I first determined these frequencies I had

[7] Although the ordinary normal modes of vibration of a tuning fork are inharmonic, it is nevertheless known that the octave may occur (See Helmholtz, Sensations of Tone, tr. Ellis, p. 54; Rayleigh, Theory of Sound, ed. 2, vol. 2, p. 463; Barton, Text-Book on Sound, arts. 309–314; Richardson, Sound p. 114). I had not supposed that the octave could be obtained with sufficient intensity to produce the above effect. The suggestion that it might be possible was made in a letter written to me by Professor A. Wilmer Duff.

[8] I have actually carried out this determination in connection with the fifth partial in only one case. But see the next paragraph with regard to the seventh partial.

no fork that went as high as the seventh partial for any bell in the Dorothea Carlile Chime except the largest. Later another fork was made for me, and the frequencies of the seventh partials found with this new fork agreed excellently with those already found by the beats of forks an octave lower.

When using this method the tone heard has the pitch of the fundamental of the fork, and it is that tone, rather than the one an octave higher, that sounds as if it were beating. This fact, and the possibility of making satisfactory determinations of the seventh partial in this way, seem to be good evidence that when a fork brings out the strike note of a bell it is really because the fifth partial is being elicited. It would seem then that this fifth partial heard along with the fundamental of the fork gives the impression of the strike note.

If this last statement is correct it seems to be additional evidence that the strike note of a bell is determined, except for octave, by the fifth partial.

OTHER TESTS

Listening at a Distance.—If the fifth partial is sufficiently prominent to determine the note heard from a bell it seemed that this partial might still be prominent when the bell is heard from a distance. I made a few attempts to find out whether the pitch from a bell was the same when I was so far away that the sound was very faint. These attempts were few and not conclusive. The pitch seemed, however, not to be affected by distance.

I spent some time listening at a distance of about three hundred meters. The bells were struck on the soundbow, and in most cases by the regular clapper. I would adjust a fork to beat with one of the partial tones of a bell, give a signal to have that bell struck, and listen for the beats. The results seemed to have most significance when the bells were struck so gently that I could just detect the beats. The strength of blow that was necessary for the hearing of beats differed for the different partials.

The results are not as definite as I had hoped, but seem to be as follows: The fourth and sixth partials are not detected when other partials can be. This is to be expected from the fact[9] that the fourth and sixth partials are not brought out strongly when the bell is struck on the soundbow. The second partial can be detected from a distance

[9] I have pointed this out on p. 258 of reference 3.

when the first cannot. The third, fifth, and seventh are the most easily detected, and the fifth perhaps more easily than the third or the seventh. A number of times I heard the fifth partial of a bell begin to beat with the fork when the bell was struck so gently that the beats were the only evidence I had that the bell had been struck—I did not hear the stroke of the bell at all.

Time of Contact of Clapper.—An attempt was made to determine the time of contact between clapper and bell. This was to be accomplished by sending an electric current through an oscillograph, the circuit being closed during the time that the clapper was in contact with the bell. On most of the bells the electric connection between clapper and bell proved to be already so good that the blow of the clapper did not produce any marked change in resistance. An attempt was made to put over the clapper a thin insulating coating and then a thin layer of metal. But the compressibility of the material added is probably too great for the results obtained to be of any value.

The largest bell of the chime is provided with three clappers. Two of these are electrically insulated from the rest of the bell, and so needed no additional insulation. Making use of one of these clappers four satisfactory records were obtained. The times of contact were found to be: 700, 820, 760, and 830 microseconds. The average of these, 780 μ sec, is about 0.28, 0.48, and 0.72 of the period of the third, fifth, and seventh partials of this bell. That is, on this bell the clapper was in contact for about half the period of the fifth partial. The data are too meager to permit of drawing any general conclusions, but if the clapper is usually in contact for about half the period of the fifth partial, it would seem likely that the development of the fifth partial would be favored. The question is to be examined further during the coming summer.

Combination Tones

It was early suggested[10] that the strike note of a bell might be a combination tone. In my 1920 paper I listed those first order difference tones which might arise from the partials I had observed and which were within a half step from the strike note. With the more accurate determinations of frequency that I now have it seems worth while to examine this question again. Among the first seven partials of the six largest bells of the Dorothea Carlile Chime I now find that the only

[10] P. J. Blessing [Physikal. Zeitschr. **12**, 597 (1911)] states that Rudolph König made this suggestion.

first order difference tones which lie within a half step from the octave below the fifth partial are these: The difference tone from the fourth and sixth partials on the *f* and *a* bells, and the difference tones from the fifth and seventh partials on all six bells. Since the fourth and sixth partials are neither of them brought out strongly when a bell is struck on the soundbow, and since the difference tones from the fourth and sixth partials are within the given range on only two of the bells, it seems that the difference tone from the fifth and seventh partials is the only one that need be considered. The difference tones from these partials are shown in Table II. On the average the difference tone from the fifth and seventh partials differs from the octave below the fifth partial by about a quarter of a half step.

TABLE II.

FIRST ORDER DIFFERENCE TONES FROM FIFTH AND SEVENTH PARTIALS

Bell	Frequency of Octave below Fifth Partial = A	Cycles/sec to add to A to get Dif. Tone	Cycles/sec to add to A to raise Pitch one Half Step
eb	305.6	4.8	18.1
f	345.5	6.5	20.5
g	387.2	2.6	23.0
ab	411.0	12.0	24.4
a	438.4	6.3	26.1
bb	462.0	− 1.0	27.4

For frequencies in the neighborhood of 300 or 400 cycles/sec Knudsen[11] finds that the ear can detect changes in pitch of something like a twentieth of a half step. On the other hand Guttmann[12] finds that singers and violinists do not attain this accuracy. He finds that the pitch of a note may be regarded as satisfactory if it is not in error by more than one per cent, which is about a sixth of a half step. If Guttmann is right the difference tone from the fifth and seventh partials on the six largest bells of the Dorothea Carlile Chime may be regarded as agreeing moderately well with the strike note.

In my work on the Harkness Chime[4] I found that the frequencies of the fifth, seventh, and tenth partials have ratios not far from 2:3:4, so that a difference tone from the seventh and tenth partials would also lie not far from the strike note. On the Dorothea Carlile Chime I have

[11] Vern O. Knudsen, Phys. Rev., **21**, 84 (1923). The value given in the text is calculated from the curve in Knudsen's Fig. 10.

[12] Alfred Guttmann, Zeitschr. f. Physchol., **58** [2d abt.], 209, 247 (1927).

not been able to determine the frequency of the tenth partial except on the two largest bells. On these two the relative frequencies of the fifth, seventh, and tenth partials are respectively: 2:3.01:4.17 and 2:3.01:4.13. On these two bells the tenth partial seems to be too far from the double octave to contribute much to the strike note.

It may be that a difference tone from the fifth and seventh partials is of some importance in determining the strike note. On the other hand, difference tones are not usually heard unless the combining tones are fairly loud. The strike note of a bell can be heard when the bell is tapped very gently.

Conclusion

The work described in this article seems to indicate that the strike note of a bell is an octave below the fifth partial, and that the fifth partial is probably important in producing the strike note. It does not settle the question as to whether a difference tone from the fifth and seventh partials may be of importance, although that seems rather doubtful, and it gives no clear answer to the question why the pitch of the bell is judged to be a octave lower than the fifth partial.

I wish to express my gratitude to the seven musicians who gave me their judgments as to the pitches of the strike notes. They were: Dr. Frank E. Dow, Professor Rebecca W. Holmes, Professor Cary F. Jacob, Professor Wilson T. Moog, Mrs. William Allan Neilson, Mr. George C. Vieh, and Professor Roy D. Welch. I wish also to record my hearty thanks to Professor George W. Alderman, of the Department of Physics at the Massachusetts Agricultural College, who worked with me for a part of the summer. We have begun some further studies, but the results are not sufficiently definite for any report as yet. We intend to to continue the work next summer.

December, 23, 1929.

THE EFFECT OF TEMPERATURE ON THE PITCH OF A BELL

By Arthur Taber Jones

Smith College

Introduction

So far as I know, the only published statement as to the effect of temperature on the pitch of a bell is contained in a paper which I wrote[1] some ten years ago. At that time my determinations of frequency were not sufficiently accurate to permit the drawing of any conclusion except that the effect must be small. A very rough theoretical calculation indicated that the pitch might fall as the temperature rose, and to the extent of perhaps a quarter of a cent[2] for each degree centigrade. That would mean that a change from midwinter to midsummer might lower the pitch of a bell by an eighth of a half step—that is, less than one per cent.

With a set of tuning forks that I now have it is possible to make more accurate determinations of pitch, and so to investigate the effect of temperature on the pitches of bells.

Method of Study

During an investigation last summer[3] on the bells of the Dorothea Carlile Chime a thermometer was frequently hung near the bells while the frequencies of their partial tones were being determined. The results on a number of bells seemed to be consistent in indicating a fall in pitch with rise in temperature. One bell was therefore selected for more careful study, and the frequencies of its first, third, and fifth partials were determined on warm days and on cool days. The three tuning forks that were used for this work were sent to the Bureau of Standards, where they were rated at temperatures near 0°C and 30°C. It was then assumed that the change in pitch of the forks is a linear function of temperature, and corrections for this change were applied to the pitches from the bell.

[1] A. T. Jones, Phys. Rev., **16,** 247 (1920).

[2] The "cent" is a unit of interval introduced by Mr. Alexander Ellis, and used throughout his translation of Helmholtz's *Sensations of Tone.* An octave contains 1200 cents.

[3] A. T. Jones, Journ. Acoust. Soc., **1,** (1930).

Results

It is known that the pitches of tuning forks fall to a slight extent with rising temperature. Before the frequencies from the bell were

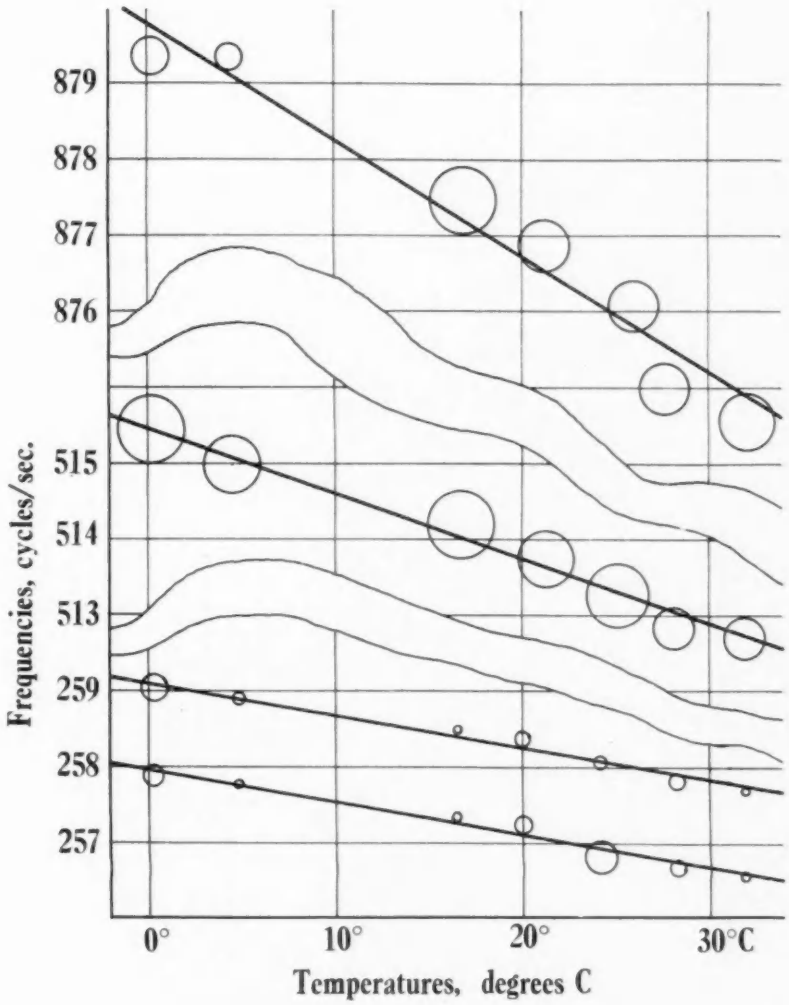

Fig. 1. *Frequencies of the first, third, and fifth partials of the a bell in the Dorothea Carlile Chime at different temperatures.*

corrected for this effect they, too, showed a small but perfectly definite change in the same direction. The correction because of the change in pitch of the tuning forks increases the amount of this effect in the bell.

The corrected frequencies of the three partial tones at different temperatures are shown in Fig. 1. Each plotted point is the average of from four to fourteen different determinations made in fairly rapid succession and with several different settings of the fork. The radii of the circles give the average deviations of these values. The first partial is a doublet, so that there are four curves instead of three.

Let us write

$$N = N_0(1-\alpha t), \tag{1}$$

where N and N_0 stand for the frequencies at t°C and at 0°C, and α is assumed to be constant. Then from the slopes of the curves in Fig. 1 the values obtained for α, beginning with the highest curve, are respectively: 0.000174, 0.000168, 0.000163, and 0.000166. The average of these is

$$\alpha = 0.000168 \text{ per degree C.}$$

The last digit in this number is probably without much significance.

From this value for α it may be seen that a change in temperature from -20°C to $+30$°C would lower the pitch by about 14 cents, i.e. by about a seventh of a half step. The effect is certainly small, and, in view of the lack of an accurate value for the temperature coefficient of Young's modulus the agreement with the rough theoretical value[1] of 0.00014 is as good as could be expected.

If we may assume that the equation (1) in reference 1 is correct we may use the above experimental temperature coefficient of frequency to calculate the temperature coefficient of Young's modulus for bell metal (Cu 78% and Sn 22%). The result is -0.00035 per degree C. This value may be interpolated satisfactorily among results obtained by Kiewiet[4] for the temperature coefficients of various copper-tin alloys.

This study has been carried out on only one bell, and perhaps no general conclusions ought to be drawn.

DECEMBER 30, 1929

[4] J. Kiewiet, Gött. Inaug.-Diss. (1886). I have not seen Kiewiet's dissertation. The values which I have used are quoted in Winklemann, Hdbuch d. Physik, ed. 2, vol. 1, p. 567 (1908).

THE MONAURAL PHASE EFFECT WITH PURE BINARY HARMONIES I. FREQUENCY RATIO 2:3

By WILLIS BEASLEY

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Although the view that quality of tone is independent of the phase relations obtaining between the partial frequencies in a complex sound has been generally accepted for a number of years, a critical analysis of the experimental work purporting to justify this conclusion warrants the objection that no evidence is available for the defense of this thesis or its antithesis.[1] In this review, the experimental records of Helmholtz (1877), König (1881), Hermann (1894), ter Kuile (1902), Lindig (1903), Lloyd and Agnew (1909), and Hartridge and Cosens (1921) are examined in detail. In any case that a demonstrable phase shift occurred, there was invariably a discriminable change in clang quality. Each investigator, however, suggests that such changes are to be explained by *assuming* periodic reinforcement and interference between identical frequencies (*Einklänge*) common to the two generators used as sources for the fundamentals; and that the periodic variation of wave form consequent upon a change in the phase relations of the fundamentals is irrelevant for hearing.

Three cogent objections appear valid: (a) the presence of such pertinent interference effects was not objectively demonstrated to be associated temporally with the audible changes in tone quality but was deduced from familiar mathematical considerations; (2) the exact nature of these heard variations was not described;[2] (3) acceptable proof of the irrelevance of phase depends on producing a condition in which the phase shift occurs under conditions favorable to observations and there are *no concomitant discriminable variations in clang quality*.

In a former paper,[3] the crucial factors essential to a solution of this problem were outlined. They are, briefly:

[1] Cf. Beasley, W. C., Experimental and Theoretical Treatment of Monaural Phase Discrimination. *Amer. J. Psychol.*, 1930 (in press).

[2] With the exception of ter Kuile, who worked with mistuned tertiary harmonies using tuning forks.

[3] Beasley, W. C., The Influence of Phase on Tone Quality. Read at the December (1929) meeting of the Acoustical Society of America in Chicago. Abstract in the *J. of the Acous. Soc. of Amer.*, 1930, *1*, 170.

(1) A derivation and production of the specific conditions under which a maximum change in compound wave form due to phase alteration between two fundamentals for a variety of frequency ratios will occur. The case of binary harmonies offers the simplest conditions for auditory analysis.

(2) This critical phase shift should be accomplished in a period of time psychologically favorable to an observation of any change that may occur; preferably time periods parametrically adjustable for each frequency ratio, and definitely measurable.

(3) An examination under these conditions of the effect when harmonics of identical frequency and known magnitude are present and phase shifts between the fundamentals and the harmonics occur simultaneously.

(4) A comparative analysis of the effect when the fundamentals maintain a *constant* phase relation, but that between the harmonics of identical frequency is altered under two conditions:[4] (a) when these harmonics are separately at various sensation levels, and (b) when they are demonstrably of sub-threshold magnitude.

(5) An analysis of the effect when all harmonics are demonstrably of sufficient sub-threshold magnitude as to be irrelevant for hearing; that is, when only two pure fundamentals are present between which the phase is altered cyclically.[5]

The apparatus shown in Fig. 1 was used as the frequency source, and since it has been described in detail elsewhere (*a.* in foot-note 5) only factors essential to the present experiments will be mentioned

[4] In this test it was, of course, necessary to use separate sources for the fundamentals and the harmonics of identical frequency.

[5] Treatment of results obtained with reference to the first four objectives will be found in the following articles by the writer to appear during 1930: a. An Improved Model of the Tone Phaser. *Amer. J. Psychol.*; b. Monaural Discrimination of Phase with Complex Sounds. *J. of Gen. Psychol.* In order to execute the fifth objective, it was necessary to obtain access to measurement devices unavailable at Ohio State. The writer wishes to acknowledge appreciation to Dr. Harvey Fletcher of the Bell Telephone Laboratories, New York City, for providing the necessary assistance in the completion of this portion of the problem; and to express specific indebtedness to the following members of Dr. Fletcher's department for aiding in certain details of the work: Dr. J. C. Steinberg for supervising the entire program of tests; K. P. Seacord for his thorough attention to perfecting circuit details; B. E. Behrens for making an electrical harmonic analysis of the tone phaser output; C. F. Sacia and G. V. Riley for photographing wave forms under a variety of conditions; and many others who contributed in a minor way from time to time. The writer does not mean to suggest, however, that any of these persons should be held accountable for any of the conclusions reported in this paper. Gratitude is also extended to Dr. A. P. Weiss and to the Graduate School at Ohio State for subsidizing this project.

here. This generator is of the rotor-inductor type, having five separate frequency outputs at levels of 750, 1350, 1500, 2250, and 2400 cycles, when the speed of the shaft common to the five rotor units is 1800 r.p.m. The frequency outputs are as constant as the cyclic rate of the power driving the motor. The line used in these tests usually varies not more than 0.42 per cent from 60 cycles during an hour's time. The speed of the motor is not altered by line voltage fluctuations.

Each rotor unit is supplied with two inductor units, which are mounted on rigid fiber discs. These discs rotate on a set of brass bushings having a center identical with that of the rotors. The inductor units of the same frequency output may be used in parallel or series with each other, or a single unit may be used. Since the rotor units are mounted with a ringing fit on a common carbonized steel shaft, the frequency

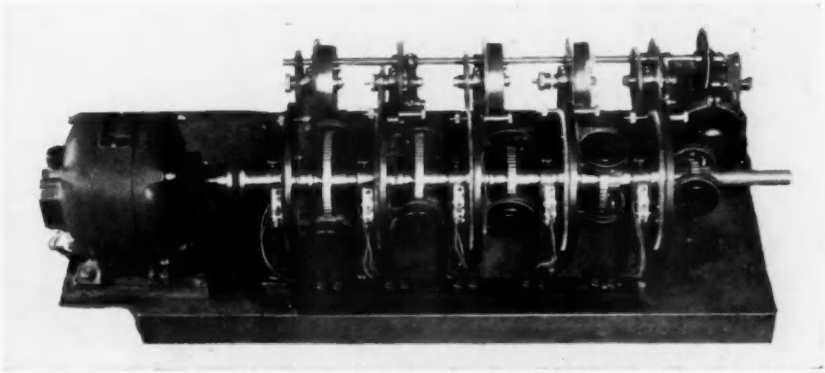


Fig. 1—The Tone Phaser.

ratio between a pair of currents remains *absolutely* constant when the corresponding inductor discs are stationary. The period of commensuration for two frequencies is determined by the *relative* number of poles on the rotors. To change the phase relations between a pair of currents, one inductor disc is maintained stationary and the other is rotated through the desired phase angle at any suitable rate. Thus the phase obtaining between the two currents can be altered continuously by infinitesimal steps during any desired period of time. This type of control, although indispensable to a solution of the problem under consideration, was not used in any former investigations.

In view of the impossibility of constructing this device so that the inductor and rotor systems have identical mathematical centers, one would expect even with the most perfect mechanical workmanship

a slight eccentricity in the path of the magnetic poles of the inductors with reference to the periphery of the rotors. If this variation is significantly large, a change in current would result during phase advance for a single unit. Such a condition might produce a discriminable variation in the relative loudness of the two tones in combination.

An analysis was made to determine the magnitude of this factor, and the data appear in Table I. Since each rotor is supplied with two inductor units, these are designated as A and B in the table. For each measurement, the inductor disc was rotated slowly until the thermocouple reading was at a minimum; then it was rotated through 180° where the current reached a maximum. The values under I_0 are the minimum and those under I are the maximum values of the current into 600 ohms in each case. The table is otherwise self-explanatory.

TABLE I

CURRENT OUTPUT OF TONE PHASER AND ECCENTRICITY OF INDUCTOR UNITS

Freq.	Unit	I-Mils		I/I_0	db Gain
		I_0	I		
2400	16 A	1.28	1.56	1.219	1.72
	16 B	.84	1.08	1.286	2.19
	16 A+B Parallel	1.56	1.66	1.064	0.54
	16 A+B Series	1.44	1.50	1.042	0.36
2250	15 A	.62	.82	1.322	2.43
	15 B	.80	1.08	1.350	2.61
	15 A+B Parallel	1.10	1.12	1.018	0.16
	15 A+B Series	1.04	1.10	1.058	0.49
1500	10 A	.92	1.30	1.413	3.00
	10 B	.49	.84	1.714	4.68
	10 A+B Parallel	.80	.86	1.075	0.63
	10 A+B Series	.86	.92	1.069	0.58
1350	9 A	.64	.68	1.063	0.53
	9 B	.84	.86	1.024	0.21
	9 A+B Parallel	.86	.86	1.000	0.00
	9 A+B Series	.94	.96	1.021	0.18
750	5 A	.84	.92	1.095	0.79
	5 B	.40	.44	1.100	0.83
	5 A+B Parallel	.88	.92	1.045	0.38
	5 A+B Series	.98	1.00	1.020	0.17

The advantage gained by using two units either in series or parallel for a single frequency is apparent from a comparison of the four values for each frequency under the heading "db Gain."

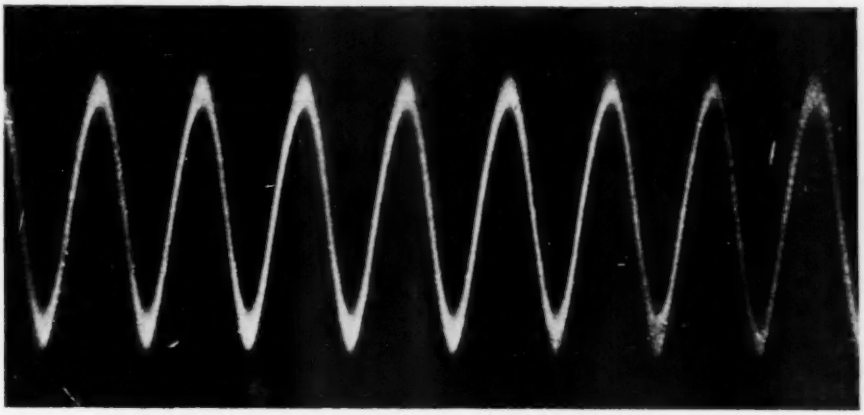


Fig. 2—Oscillogram of Tone Phaser Current Wave.

The wave form of the current output of the tone phaser closely approximates sinusoidal curvature, as shown in Fig. 2. This oscillogram was recorded by means of a General Electric mechanical oscillograph. In comparing this to the wave form of Fig. 3, one notes that the tone phaser wave has no harmonics of appreciable magnitude; and especially

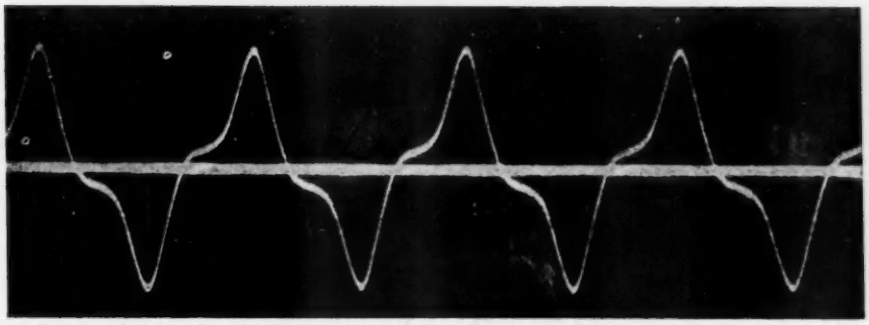


Fig. 3—Typical A.C. wave with strong third harmonic (from Miller, *Science of Musical Sounds*, by permission of the author).

significant is the fact that the third harmonic is not present in this wave form—a deviation from the usual character for this type of generator. However, the current wave may contain harmonic frequencies of too

small a magnitude to be recorded by the oscillograph. Consequently, it appeared necessary to obtain an electrical harmonic analysis of the current for each unit on the tone phaser. This was accomplished by means of a modulating harmonic analyzer.[6]

TABLE II

ELECTRICAL HARMONIC ANALYSIS OF TONE PHASER OUTPUT

Fundamental Frequency c.p.s.	Harmonic Loss (decibels)									
	2		3		4		5		6	
	Unf.	F.	Unf.	F.	Unf.	F.	Unf.	F.	Unf.	F.
750	1500		2250		3000		3750		4500	
	24	*52	43	*52	*52	*52	*52	*52	*52	*52
1350	2700		4050		5400					
	28	*52	*52	*52	..	..	..	..	..	..
1500	3000		4500		6000					
	26	*52	44	*52	..	..	..	..	..	..
2250	4500		6750							
	29	*52	..	..	..	..	..	..	..	..
2400	4800		7200							
	30	*52	..	..	..	..	..	..	..	..

* Means that the harmonic loss was greater than 52 db, but could not be measured.

The data obtained from this analysis are recorded in Table II. The efficient frequency range of this analyzer is from 60 to 5000 cycles. In order to evaluate the magnitude of harmonics an arbitrary intensity level is used for the fundamental, and subsequently the comparison level of each harmonic is evaluated in decibels loss with reference to the intensity of the fundamental. The table shows this *db* loss for both the filtered and unfiltered output of the tone phaser for each harmonic within the frequency range of the analyzer and for each fundamental generated. For example, in the harmonic series for the fundamental of 750 cycles, the second harmonic (1500 cycles) was 24 *db* less than the fundamental when the current was unfiltered, and greater than 52

[6] Cf. Moore, C. R., and Curtis, A. S., An Analyzer for the Voice Frequency Range. *Bell Sys. Tech. Journal*, 1927, *6*, 217–229.

db less than the fundamental when band pass filters were in the circuit. In each case that filters were used, the high pass filter was set at from 150 to 250 cycles below the fundamental, and the low pass filter was set at from 150 to 250 cycles above the fundamental. The filters always attenuated harmonics somewhat more than 28*db*. The relative sensitivity of the analyzer in these tests for direct measurement is about 52*db*; that is, a harmonic suppressed more than 52 *db* below the fundamental could not be measured directly, although it could be definitely classified as being below this order of intensity.

In the tests to be described later harmonics were at least 52*db* below the fundamental where the band pass filters were used in the circuit. An interesting fact to note is that in the harmonic series for the fundamental 750, where the first six harmonics are within the frequency range of the analyzer, there is practically a logarithmic decrement series in the intensity of harmonics, regardless of odd or even numbered terms. With this type of generator there is usually an exceptionally strong third harmonic, and all the even numbered harmonics are either absent or exceedingly weak in comparison to the odd numbered ones. The assumption appears valid that any harmonic frequency greater than 5000 cycles would be even more than 52 *db* below the fundamental, and consequently too far sub-threshold for hearing when the fundamental is used at a sensation level not to exceed 30 decibels.

For the problem of phase discrimination monaurally it is of fundamental importance to attenuate all harmonics of the same frequency below the threshold for hearing; otherwise, when the phase is altered between the fundamentals, that between the identical frequencies is simultaneously altered. This produces a periodic augmentation and diminution in the amplitude of a single frequency, becoming manifest in the clang as a loudness variation in one or more simple tone components.

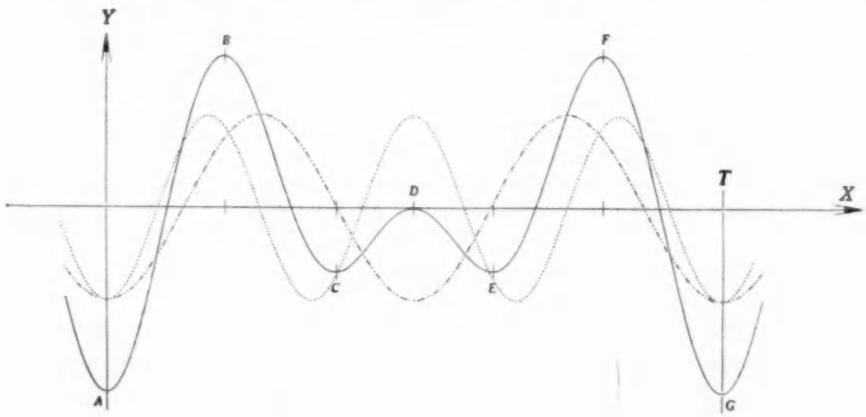
The frequencies of importance in this problem may be identified readily from Table III. For example, in case of the combination 1500 and 2250 cycles (frequency ratio 2:3), the harmonics of 4500, 9000, 13500, 18000, and 22500 cycles are common to both systems. If the phase of one fundamental is advanced at a constant rate, then each of these higher frequencies would change in amplitude periodically. However, the period of change for each successive harmonic would diminish arithmetically by increments of unity; that is, during one period of change for the 4500 cycle harmonic, the 9000 would change twice,

TABLE III

HARMONIC SERIES OF THE FIVE TONE PHASER FREQUENCIES
(First Sixteen Terms)

Term	Frequencies				
1	2400	2250	1500	1350	750
2	4800	4500	3000	2700	1500
3	7200	6750	4500	4050	2250
4	9600	9000	6000	5400	3000
5	12000	11250	7500	6750	3750
6	14400	13500	9000	8100	4500
7	16800	15750	10500	9450	5250
8	19200	18000	12000	10800	6000
9	21600	20250	13500	12150	6750
10	24000	22500	15000	13500	7500
11	26400	24750	16500	14850	8250
12	28800	27000	18000	16200	9000
13	31200	29250	19500	17550	9750
14	33600	31500	21000	18900	10500
15	36000	33750	22500	20250	11250
16	38400	36000	24000	21600	12000

the 13500 three times, the 18000 four times, and the 22500 five times. On the tone phaser there is an automatic recording device which registers the rate for the first pair of identical frequencies in each system.



RATIO 2:3 *AMPLITUDES* 1:1 *PHASE* 1

Fig. 4—One characteristic period of the function $y=\sin(3x+270°)+\sin(2x+270°)$.

Reacting to a periodic change in a component of the clang by pressing a key and recording on a moving tape the rate of these discriminations parallel to the record of the harmonic rate of change, one can ascer-

tain readily if any of these harmonics are present to an audible degree. Under the conditions of our test these harmonics were definitely rendered sub-threshold, so that this effect did not enter into the observations.

The brass gear devices on the tone phaser (Cf. Fig. 1, *supra*) are designed to produce a cyclic change of phase between the fundamentals in a known period of time, which ranges from two to twelve seconds. From a consideration of the curves in Figs. 4, 5, and 6, the type of variation in compound wave form produced by shifting the relative phases of the fundamentals is apparent.

We select these three phase relations arbitrarily for illustration. Phase 1 (Fig. 4) is one characteristic period of the function

$$Y = \text{Sin}\ (3x + 270°) + \sin\ (2x + 270°) \quad , \qquad (1)$$

and may be called the phase of coincident minima, since the two waves pass through the origin at a phase of 270.° In equation (1) as in the subsequent ones, the phase angle notation specifies the phase of that component at the origin.

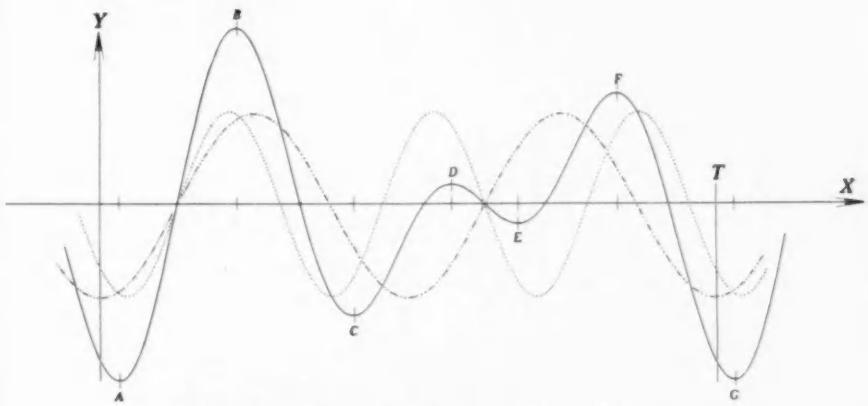


Fig. 5—One characteristic period of the function $y=\sin\ (3x+225°)+\sin\ (2x+270°)$.

Phase 2 (Fig. 5) is one period of the function

$$Y = \sin\ (3x + 225°) + \sin\ (2x + 270°) \qquad (2)$$

and is called the phase of opposition, since during one periodic time of this function there are neither coincident minima nor maxima, but a condition mid-way between these two extremes is maintained. In this phase relation the $3x$ component has been retarded by 45° with reference to its phase at the origin in phase 1.

Phase 3 (Fig. 6) is one period of the function

$$Y = \sin (3x + 180°) + \sin (2x + 270°) \qquad (3)$$

and is called the phase of coincident maxima. In phase 3 the $3x$ component has been retarded by 45° with reference to its phase at the origin in phase 2, and 90° with reference to phase 1.

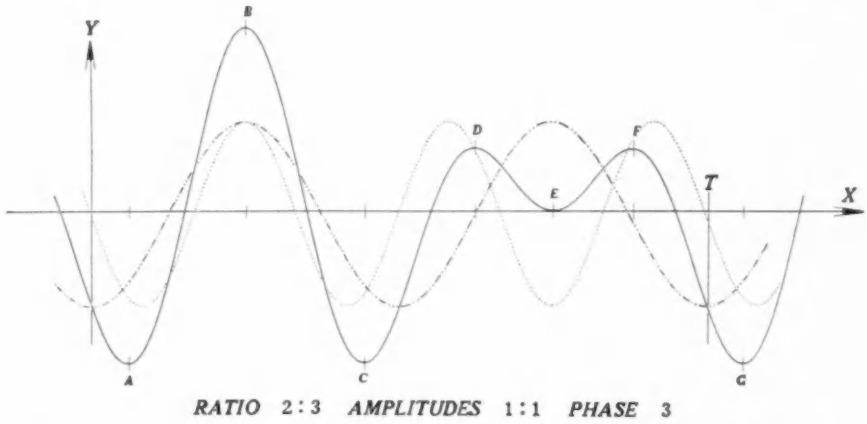


Fig. 6—One characteristic period of the function $y=\sin (3x+180°)+ \sin (2x+270°)$.

The points of change in direction (not sign), marked on the trigonometric resultant curve A, B, C, D, E, F, and G, are evaluated by analytic solution in Table IV. These abscissae values are accurate to one minute.

TABLE IV

DISPLACEMENT VALUES FOR MAXIMUM AND MINIMUM POINTS ON RESULTANT CURVE: RATIO 2:3, PHASES 1–3, EQUAL AMPLITUDES
(For Equations 1–3. *See Text*)

Max. and Min. Pts.	Phase 1		Phase 2		Phase 3	
	Abscissae	Ordinates	Abscissae	Ordinates	Abscissae	Ordinates
A	0°	−2.00	10°25′	−1.91	21° 5′	−1.63
B	68°55′	1.63	79°35′	1.91	90°	2.00
C	133°55′	−0.71	147°25′	−1.22	158°55′	−1.63
D	180°	0.00	205°20′	0.22	223°55′	0.71
E	226° 5′	−0.71	244°40′	−0.22	270°	0.00
F	291° 5′	1.63	302°35′	1.22	316° 5′	0.71
G	360°	−2.00	370°25′	−1.91	381° 5′	−1.63

The absolute displacement values of phases 1 and 3 are equal, but opposite in sign; that is, phase 3 is an inversion of phase 1. This can be noted by either comparing the values in Table IV, or by inverting Fig. 6 in relation to Fig. 4. Phase 2, however, yields not only a differences in *form*, but also a diminution in the *gross amplitudes* of the points of change on the resultant curve. For example, the maximum amplitude for phase 1 is -2.00 (points A, B, and G); but for phase 2 is -1.91 (points A, B, and G), constituting a diminution of 0.09 units or 4.5 per cent of the maximum amplitude in phase 1. The maximum amplitude in phase 3 is again 2.00 as in phase 1 (the change in sign here is irrelevant, since we are dealing with absolute displacement values). A second characteristic change is the diminution and increase in various individual displacements of the periodic wave. With reference to phase 1 in each of the following instances, point A diminishes 4.5 per cent in phase 2, and 18.5 per cent in phase 3; point *F* diminishes 25.1 per cent in phase 2, and 56.4 per cent in phase 3. Other changes of this character may be noted from the table (compare also Fig. 8, *infra*.).

A change from phase 1 through phase 3 and back to phase 1 is termed a *cyclic* phase shift. For the ratio 2:3 a retardation in the $3x$ component of 180° yields the same condition as equation (1); consequently, a continuous change in phase from the conditions of equation (1) to those of the equation

$$Y = \sin (3x + 90°) + \sin (2x + 270°) \qquad (4)$$

constitutes a cyclic phase shift. The gear mechanism on the tone phaser can be set to produce this shift in either 2, 4, 6, 8, 10, or 12 seconds. A phase contact breaker indicates automatically this period of time, and is independent of the generating system supplying current to the transmitters. This gear mechanism is complicated due to the fact that the amount of phase retardation in the component of higher frequency necessary to produce a cyclic phase shift varies for different frequency ratios.

These amounts may be calculated readily from the expression

$$Y = \sin\left[HX + \left(270° - \frac{90(P-1)}{\mathrm{L}}\right)\right] + \sin (LX + 270°) \qquad (5)$$

In equation (5) H and L are the ratio terms for the components of higher and lower frequency respectively; and P is the particular *phase condition* desired, having a corresponding numerical value. Thus, five

phase conditions are specified for P in this formula. Phases 1, 2 and 3 have been described. Phase 5 yields the same condition as phase 1, and phase 4 a condition mid-way between phases 3 and 5. (Cf. the first, third, fifth, seventh and ninth waves in Fig. 8, *infra*, which correspond to phases 1 to 5).

Table V contains values calculated from equation (5) for several frequency ratios, to illustrate the dependence of the amount of phase retardation on the frequency ratio in producing the maximum changes in compound wave form due to phase shift.

TABLE V

DEGREES OF PHASE RETARDATION IN H-COMPONENT TO OBTAIN SPECIFIED PHASE CONDITION

Phase Cond.	Frequency Ratio					
	1:2	2:3	3:5	5:8	9:10	15:16
1	0	0	0	0	0	0
2	90	45	30	18	10	6
3	180	90	60	36	20	12
4	270	135	90	54	30	18
5	360	180	120	72	40	24

It is seen that, for example, to obtain phase 3 compared to phase 1, the component of higher frequency must be retarded 180° for the ratio 1:2, 90° for the ratio 2:3, 60° for 3:5, 36° for 5:8, 20° for 9:10, and 12° for 15:16, etc. The application of equation (5) to experimental procedure in this problem appears crucial, since the reaction of a receptor is inversely proportional to the square of the time and approximately directly proportional over medium levels to the logarithm of the increment in stimulus magnitudes. The *duration* of a stimulus change is, consequently, of as paramount significance as the *magnitude* of the increment for the discriminatory characteristics of the organism. Former experiments altogether neglected this factor.

Oscillograms made of the current output of the tone phaser, using frequencies 1500 and 2250 in combination (see circuit diagram, Fig. 9), are illustrated in Figs. 7 and 8. Fig. 7 shows a cathode ray oscillogram for the ratio 2:3, of phase conditions closely approximating phases 1, 2 and 3. These conditions were maintained constant for about two minutes for each phase relation. By comparing these wave forms to those of Figs. 4, 5 and 6, one can note the degree of correspondence. A more revealing illustration is presented for inspection in Fig. 8.

which represents a cyclic phase change for the ratio 2:3. This oscillogram was made with a high frequency mechanical oscillograph, which utilizes a motion picture film moving continuously for recording suc-

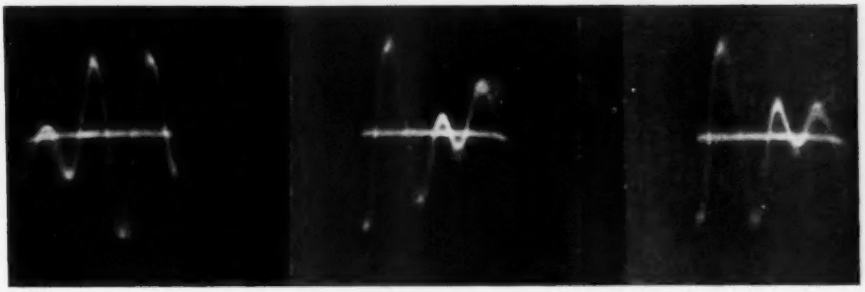


Fig. 7—Cathode ray oscillogram of phase conditions 1, 2 and 3 for the ratio 2:3. Equal amplitudes of the fundamentals.

cessive changes in the wave form. For this record the phase of the higher frequency component was retarded continuously at the rate of one cyclic phase change in about five seconds. A continuous record was made of the change in wave form occurring during this five seconds. Seg-

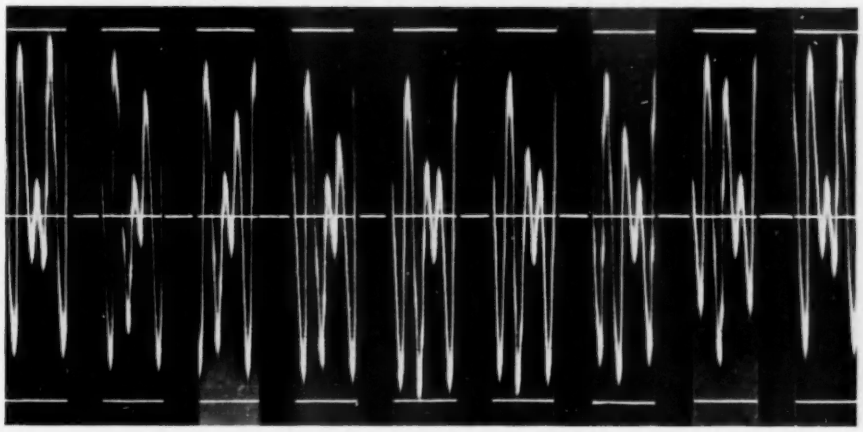


Fig. 8—One cyclic phase shift for the frequency ratio 2:3, equal amplitudes for the fundamentals. The initial and final stages are phases 1 and 5.

ments of this film were selected to illustrate the successive periods of change as shown in Fig. 8. A careful inspection of this oscillogram will make clear exactly what happens to the wave form during one

cyclic phase shift. The initial and final stages are phases 1 and 5. The third, fifth, and seventh waves are phase conditions 2, 3, and 4 respectively.

Note two characteristic changes: (a) individual displacements of the resultant wave diminish and increase *once* in amplitude throughout the cyclic period; (b) the total excursion of the maximum displacement of the resultant wave varies from maximum to minimum absolute value *twice* during one cyclic phase shift, reaching maxima at phases 1, 3, and 5, and minima at phases 2 and 4. This offers grounds for the sug-

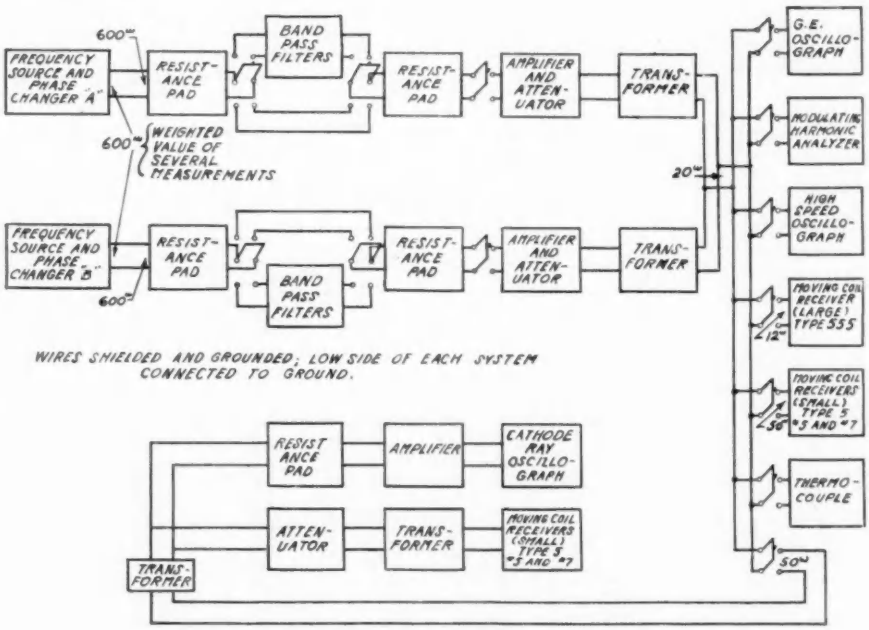


Fig. 9—Diagram of the Test Circuit.

gestion that if the monaural phase effect is of a discriminable order, we might expect a variation occuring *once* or *twice* during a cyclic phase shift, depending upon which characteristic change in the resultant wave is responsible for the audible changes; or perhaps both types of stimulus changes are discriminable, in which case one would have twice the frequency of the other. Further, in view of the data presented in Table IV, we might expect hypothetically a differential in the magnitude of these effects if both are discriminable. The essential condition is that the cyclic phase shift be produced continuously at a constant rate, with a time duration favorable to discrimination.

The circuit arrangement used is shown in Fig. 9, which is largely self-explanatory. In arranging and using this circuit, emphasis was placed on taking precautions against introducing harmonics due to over-loading, and on eliminating extraneous pick-up. All leads were shielded and grounded, and the low side of each system was connected to the ground. The headphones used in most of the observations were the small size moving coil receivers (Type 5), but in some cases the large size (Type 555) were used. These receivers can be used safely up to a sensation level of 30 decibels without over-loading and introducing harmonics not present in the actuating current, especially at the frequency levels we used (750 to 2400 cycles).[7]

Discussion of Observations

For the tests on the frequency ratio 2:3, the *A* source (Cf. Fig. 9) was 1500 cycles, and the *B* source was 2250 cycles.[8]

In arranging conditions for observations, the high pass filter in system *A* was first set to pass frequencies *above* 2000 cycles, and the low pass filter was cut out. Then the *A* frequency source was connected in, and the *B* frequency source cut out. The harmonic series for the 1500 cycle source, with the fundamental attenuated below threshold, was the only audible output of the receivers. Then the output of *A* was attenuated until the previously very faint harmonics became inaudible. This made certain that no harmonics of an audible order were present in the acoustic output of the system due to source *A*, even when the low pass filter was not in the circuit. The low pass filter was inserted, set at 2000 cycles; and the high pass filter was reduced to 1250 cycles. The fundamental of 1500 cycles could now be heard at a sensation level of about 10 *db*. With the *A* system thus calibrated, an equivalent procedure was used for system *B*. Here the final setting of the band pass filter was 2000 for the high pass and 2500 for the low pass. With this arrangement, then, in which harmonics were at least 25 *db* below threshold, and in which the fundamentals were separately at sensation levels of about 10 *db*, observations were made as to the phase effect.

The observers were seated one at a time in a sound-proof room and instructed to discriminate by the method of paired comparison between two conditions: one in which the phase relations between the fundamen-

[7] Cf. Wente, E. C., and Thuras, A. L., High Efficiency Receivers of Large Power Capacity. *Bell Sys. Tech. Journal*, 1928, 7, 140–153.

[8] The observations and illustrative data for the ratios 1:2, 1:3, 3:5, 5:8, and various combinations of three, four and five frequencies will be incorporated in a future report.

tals were being altered continuously for thirty seconds, and one in which the fundamentals maintained a steady phase relation for thirty seconds. A series of ten conditions constituted each set of observations, and each of three trained observers went through five series.

The observer marked *R* in a table if he judged that the condition to which he listened exhibited variations of any sort, and *S* if the tonal pattern appeared steady and exhibited no discriminable change. The subject was told that in each case of two conditions to compare, one condition would be continuous phase change and the other would be a steady condition. However, the observer was not informed as to which condition would be steady and which one phase changing. The order for these conditions was irregular, and was not duplicated in different series. The results of these observations are recorded in Table VI. The general average for all observations indicates only 54 per cent correct judgments.

TABLE VI

PER CENT CORRECT JUDGMENTS IN DISCRIMINATING CONDITIONS *R* AND *S* BY METHOD OF PAIRED COMPARISON

Observer	Series					Average
	1	2	3	4	5	
A	50	40	50	60	70	54
B	30	60	40	50	40	44
C	60	70	50	60	80	64
Average for all observations						54

Although this does indicate that the phase changing condition could not be discriminated consistently (any better than chance) from the steady condition, it does not mean that no discriminable changes in tone quality occurred. Each observer agreed that in both the *R* and *S* conditions very faint variations in loudness could be heard; and the difficulty encountered by each was in discriminating a difference between these changes in the two conditions. Working at this low sensation level, however, one might expect such ambiguity. These phenomena have been known to psychologists for some time, and are generally called "fluctuation of attention." This is a classic demonstration in vision by means of a Masson disc, where faint gray rings on a white background fade and increase irregularly in blackness. In listening to a harmony, such as the fifth, at a low sensation level the observer reacts

first to one component and then to another, and such a shift in "attention" results in an emphasis of the component being reacted to at the time. With a periodic recurrence of such shifts, an effect equivalent to irregularly occurring beats is produced. However, such changes are characteristically *irregular* in duration. Under a condition where the cyclic phase shift is produced at *regular* intervals, if there is a differential effect due to phase, the two conditions of *R* and *S* could best be discriminated in terms of their *comparative regularity*.[9] The apparatus necessary for making a record of discriminations of this type was not available for these observations; and so for the time being this aspect of the problem remains undetermined. We can definitely conclude, however, that at this low sensation level (10 *db*) with the combination 1500 and 2250 cycles, a pure binary harmony, the monaural phase effect is of such little consequence that it cannot be discriminated from the steady state, except (perhaps?) by special techniques which remove ambiguities due to characteristic but irregular organic variations of the observer.

If instead of using a thirty-second period for each of two conditions much shorter periods (say of two or three seconds each) are used, there is no discriminable difference at this level between the harmonies in phase 1 as compared to phase 2 or 3; that is, where we shifted instantaneously from phase 1 to phase 2 or 3, and back to phase 1 with a brief pause of silence between the successive phase conditions, each of fifteen observers could detect no differences in tone quality.

For the next set of observations on this frequency ratio, other conditions remaining the same, the level was raised 20 *db*. This was about the maximum intensity that could be used without raising the level of harmonics above threshold. At this sensation level (25 to 30 *db*), the same observers recorded in Table VI were able to discriminate on the average between the *R* and *S* condition with 90 per cent correct judgments. At this level the "fluctuation of attention" factor was not so confusing as in the former case, although still present to a slight degree; but the *R* condition appeared to introduce a sufficient variation in loudness at this level to enable the observers to discriminate between the *R* and *S* conditions.

Six other observers who had not served in the former situation could report no change in tone quality in comparing the *R* and *S* conditions

[9] The importance of using a technique whereby an observer can discriminate two conditions of change on the basis of duration is illustrated clearly in the initial experiments reported in article b. foot-note 5, *supra*.

at a sensation level of 25 to 30 *db*, but when the level was raised to about 40 *db*, each reported at once variations in the relative loudness of the components of the clang. It was the consensus of opinion among these observers that there were objective harmonics now in the clang, although they certainly were not present due to the current constituents. It is probable that at this intensity level the receivers began to overload, and objective harmonics were introduced. The moment objective harmonics appeared, the difference could be readily detected between the *faint* loudness surge due to phase shift, and the interference effect due to harmonics.

In correlating these observations with the two types of variation in wave form mentioned above in connection with Figs. 4, 5, 6, and 8, it appears that the phase effect is of an order associated with the 4.5 per cent change in maximum amplitude of the resultant wave, and not at all sufficiently conspicuous to be connected with the profound changes in the individual amplitudes of the compound wave amounting to magnitudes of 25.1 and 56.4 per cent.

Another set of observations were made, in which the observers could listen in the receivers and simultaneously observe visually the changes in compound wave form by means of the cathode ray oscillograph. Under such a condition of observation, none of the fifteen observers could detect an auditory variation associated temporally with the conspicuous amplitude changes in the elements of the compound wave, even with such an ideal condition for objective cues as to the rate of such changes. This was true for all levels below 25 *db*, where it was certain that objective harmonics were absent. This again refers the phase effect to maximum amplitude change, which occurs twice during one cyclic phase shift, rather than to the more conspicuous change in compound wave *form*.

THE USE OF SOUND IN NAVIGATION

By Horatio W. Lamson
General Radio Co.

The application of acoustics as an aid and safeguard to navigation is of ancient origin. Fog whistles and bells have long served their purpose as warning signals to the wary navigator blinded by fog or darkness, but when needed most they have sometimes failed him utterly. Attenuated by adverse winds and reflected by capricious fog banks, such airborne sound waves are often unreliable in their range and apparent direction of travel. The sea itself, due to its homogeneous nature and its superior conductivity of sound, provides, fortunately, a far more desirable medium for the transmission of acoustic signals.

Prior to the World War, underwater sound beacons were located with more or less success by alternately listening upon microphones located on each side of the ship in tanks below the water line and swinging the ship to give responses of equal intensity when headed towards the beacon. This was obviously a troublesome and often dangerous procedure when navigating in shoal or restricted waters.

Realizing its value as a weapon in submarine warfare, the U. S. Navy upon our entry into the War began, with the assistance of several commercial concerns and American men of science, an intensive research in the field of submarine acoustics. The introduction of the principle of binaural listening into this art crowned these endeavors with success and resulted first in the famous "C" tube, a device wherein the ears of the observer were figuratively placed upon the extremities of a horizontal beam rotated beneath the surface of the sea. Binaural observations upon the menacing "U" boats could then be made by swinging this beam until it became perpendicular to the direction of the hidden submarine.

The "C" tube was, however, somewhat cumbersome in operation and necessarily required a listening vessel to be lying to when making observations. Improvements in the art subsequently permitted the use of two fixed sound receivers mounted on the hull of the ship. This was of obvious tactical value in that it permitted observations to be made while the vessel was steaming ahead hunting its prey. In order to obtain binaural observations with fixed receivers, a device called a *compensator* was employed to introduce variable amounts of time lag or

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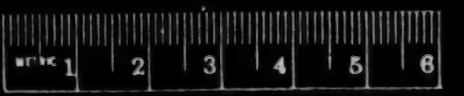


at a sensation level of 25 to 30 *db*, but when the level was raised to about 40 *db*, each reported at once variations in the relative loudness of the components of the clang. It was the consensus of opinion among these observers that there were objective harmonics now in the clang, although they certainly were not present due to the current constituents. It is probable that at this intensity level the receivers began to overload, and objective harmonics were introduced. The moment objective harmonics appeared, the difference could be readily detected between the *faint* loudness surge due to phase shift, and the interference effect due to harmonics.

In correlating these observations with the two types of variation in wave form mentioned above in connection with Figs. 4, 5, 6, and 8, it appears that the phase effect is of an order associated with the 4.5 per cent change in maximum amplitude of the resultant wave, and not at all sufficiently conspicuous to be connected with the profound changes in the individual amplitudes of the compound wave amounting to magnitudes of 25.1 and 56.4 per cent.

Another set of observations were made, in which the observers could listen in the receivers and simultaneously observe visually the changes in compound wave form by means of the cathode ray oscillograph. Under such a condition of observation, none of the fifteen observers could detect an auditory variation associated temporally with the conspicuous amplitude changes in the elements of the compound wave, even with such an ideal condition for objective cues as to the rate of such changes. This was true for all levels below 25 *db*, where it was certain that objective harmonics were absent. This again refers the phase effect to maximum amplitude change, which occurs twice during one cyclic phase shift, rather than to the more conspicuous change in compound wave *form*.

THE USE OF SOUND IN NAVIGATION

By Horatio W. Lamson
General Radio Co.

The application of acoustics as an aid and safeguard to navigation is of ancient origin. Fog whistles and bells have long served their purpose as warning signals to the wary navigator blinded by fog or darkness, but when needed most they have sometimes failed him utterly. Attenuated by adverse winds and reflected by capricious fog banks, such airborne sound waves are often unreliable in their range and apparent direction of travel. The sea itself, due to its homogeneous nature and its superior conductivity of sound, provides, fortunately, a far more desirable medium for the transmission of acoustic signals.

Prior to the World War, underwater sound beacons were located with more or less success by alternately listening upon microphones located on each side of the ship in tanks below the water line and swinging the ship to give responses of equal intensity when headed towards the beacon. This was obviously a troublesome and often dangerous procedure when navigating in shoal or restricted waters.

Realizing its value as a weapon in submarine warfare, the U. S. Navy upon our entry into the War began, with the assistance of several commercial concerns and American men of science, an intensive research in the field of submarine acoustics. The introduction of the principle of binaural listening into this art crowned these endeavors with success and resulted first in the famous "C" tube, a device wherein the ears of the observer were figuratively placed upon the extremities of a horizontal beam rotated beneath the surface of the sea. Binaural observations upon the menacing "U" boats could then be made by swinging this beam until it became perpendicular to the direction of the hidden submarine.

The "C" tube was, however, somewhat cumbersome in operation and necessarily required a listening vessel to be lying to when making observations. Improvements in the art subsequently permitted the use of two fixed sound receivers mounted on the hull of the ship. This was of obvious tactical value in that it permitted observations to be made while the vessel was steaming ahead hunting its prey. In order to obtain binaural observations with fixed receivers, a device called a *compensator* was employed to introduce variable amounts of time lag or

retardation into the paths joining the two receivers individually to the ears of the observer.

In this manner, sound waves picked up in general non-synchronously by the receivers could be made to arrive simultaneously at the two ears, producing thereby a binaurally centered sound image. Obviously the necessary amount of artificial retardation required to accomplish this purpose could be calibrated in terms of the angular bearing of the target with respect to the line of the receivers. In the acoustical system employing hollow rubber sound receivers, the compensator consisted of adjustable air columns inserted in the paths of the stethoscopes. In the later-perfected electrical system, the retardation to electrical impulses, generated by the microphone sound receivers, was accomplished by means of artificial lag lines built up of sections, each having a series inductive and a shunt capacitive element, and introduced in greater or less quantities by suitable switching mechanisms.

The type of sound receiver or *hydrophone* used with such a simple *two-spot* binaural system must obviously be equi-sensitive to sounds coming from any direction. Such a system proved excellent when one source of sound alone was present, but it was subject to considerable interference in the presence of a multiplicity of sounds, the usual condition in practice.

This difficulty has been overcome to a large degree in the modern *multi-spot systems* employing a group of hydrophones for each ear in conjunction with a multi-spot compensator. The two groups are separately focused by appropriate lag line adjustments to give a maximum response for a particular direction, while minimizing undesired sounds from all other directions. Each group then becomes virtually a single sound receiver having a *directional selectivity*. The two groups are subsequently balanced binaurally as in the two-spot system. These operations are accomplished simultaneously by a single control mechanism which may obviously be calibrated to give directly the angular bearing of the sound target focused upon with respect to the keel of the vessel. The multi-spot compensator, developed to a large extent in its modern forms by the General Radio Company, gives therefore a centered response of maximum intensity.

The hydrophones are usually arranged in two lots of 6, 12, or 18, mounted in straight lines approximately parallel to the keel on each side of the vessel near the bow. The forward half of such a line constitutes the group which is connected to the left ear, while the aft half

forms the group connected to the right ear of the observer. By means of a selector switch, the operator may listen either on the port or on the starboard side of the vessel at will. In this manner, owing to the shielding action of the hull of the ship, he can readily differentiate between the two ambiguous bearings given by any one setting of the compensator, that is, determine whether the sound source is on the port or starboard side of the vessel. The hydrophones are sometimes placed in ballast tanks in the ship's bottom and are sometimes mounted outside the ship's skin and covered with a protective stream-lined and water-filled shield known as a "blister."

The multi-spot hydrophone system has proven to be one of the many noteworthy contributions of the U. S. Navy to the art of navigation.

With this brief review of the development of the art fostered as an instrument of war, we propose to discuss its use as a peacetime aid to navigation when employed either by naval or commercial vessels.

When making a landfall after several days at sea in overcast weather permitting infrequent or perhaps no sextant observations, the navigator values highly reliable information of any sort which will assist him in checking his dead-reckoning course, and fixing his position on the chart with greater certainty. With the sun, the stars, and the shore beacons invisible, two sources of information remain; namely, observations, other than visual, upon known shore beacons, and the determination of the depth of water beneath the vessel. The radio compass is a valuable instrument for the former, while the lead line serves in the latter case. The hydrophone system fortunately may furnish information on both of these points.

Many of the lightships distributed along the coastal waters of our own and other countries are equipped with large underwater bells developed by the Submarine Signal Company. These are struck pneumatically with some distinguishing code signal whenever foggy conditions prevail. Hydrophone bearings have frequently been obtained upon these submarine bells at distances as great as a score or more miles, while the observing vessel was approaching or passing on any desired course. A single observation gives, of course, only the direction of the sound beacon with respect to the ship's keel. By a series of observations taken while the ship is traveling at a known speed on a fixed compass course, data are afforded for obtaining by simple triangulation methods the distance, as well as the direction, of the beacon at any point along the course; in other words, for fixing the location of the vessel on the chart.

When navigating carefully charted waters, a single bearing upon an underwater sound beacon, combined with a simultaneous sounding of the depth of the sea, will frequently provide the necessary data for a satisfactory "fix" of the ship's position.

Sounding the depth of the sea by means of the lead and line is ancient procedure, a laborious, and, in rough weather, often a dangerous undertaking, and one which requires reduced headway with attendant loss of time and other navigational difficulties. An invention of the Navy Department bids fair to supersede the boatswain's picturesque "Mark 7" heaving of the line. We refer to the use of the hydrophone in obtaining depth soundings by the *angle of reflection* method. The line of hydrophones, as previously stated, is located near the bow of the vessel, while the ship's propeller churning the sea serves as a strong source of underwater sound at the stern. Now, this sound travels from the propeller to the hydrophones by three principal paths: first, directly through the water along the sides of the vessel; second, by internal reflection at the air-water surface amidships; and third, by reflection from the sea bottom beneath the ship. Furthermore, since the submergence of the hydrophones and propeller is relatively small compared to their fore-and-aft separation, so that the first and second paths are nearly equal in length, and since the second sound ray suffers practically a 180-degree phase reversal on reflections, the first two rays cancel at the hydrophones, leaving essentially all the energy observed to be that reflected from the sea bottom. The degree with which this condition exists, especially for the lower frequency sounds which predominate in the noise of the propeller, is amazing. Obviously, for shallow water the path of the returning echo is nearly horizontal, while it approaches the vertical as the depth increases.

A little consideration will show that a horizontal line of hydrophones used in conjunction with a multi-spot compensator focuses not only upon sound rays traveling in a horizontal plane, but, in fact, upon any rays forming the elements of a conical surface which has a horizontal axis with the hydrophones at the apex. In other words, the compensator may be focused to measure the angle in the vertical plane made by the echo of the propeller sounds coming from the sea bottom. For a given propeller-hydrophone separation, the observed angle, that is, the compensator dial scale, may obviously be calculated to read directly in terms of the depth of the sea. The operator, therefore, has merely to focus his compensator upon the sound of his own propellers and to call

out every minute, if desired, the depth of the sea, *while the vessel is proceeding at full speed on its desired course.*

Since the depth is a tangent function of the echo angle, soundings by the *angle of reflection method* lose precision as the depth increases, but they may be taken with satisfactory accuracy up to depths equal to about three times the length of the vessel.

The lead line brings up a sample of the soil on the sea bottom for examination which, of course, the hydrophone cannot do, but an experienced operator by observing the *quality* of the echo sounds can form a good idea of the character of the bottom surface whereat they were reflected.

The art of sounding the depth of the sea by timing a sound echo from the bottom is old. A modern application of this method contributed by the U. S. Navy is a device known as the Sonic Depth Finder. This instrument is essentially an automatic transmitting key which triggers off a continuous series of short, uniformly timed sound impulses from an electrical submarine oscillator. These impulses, by nature a few cycles of a frequency of 550 or 1050 cycles per second, are radiated into the sea from the surface of a heavy diaphragm driven by electromagnetic means. The operator of the device wears a special telephone headset which enables him to hear in one ear the outgoing impulses, while in the other ear he perceives the echo impulses picked up by the hydrophone equipment. By the use of suitable attenuators, these impulses are approximately balanced in intensity. In general, the outgoing impulses and the echoes will not be synchronous. The operator now manipulates the control mechanism of the Sonic Depth Finder, varying the rate at which the impulses are transmitted until the outgoing impulses and the echoes become exactly synchronous; that is, until No. 1 echo arrives back at the ship at the exact instant when No. 2 impulse is leaving the vessel. Under these conditions, the time interval between the successive repeating impulses represents the time of transit of the sound waves to the sea bottom and back, so that the adjustment of the Sonic Depth Finder may be calibrated to read directly in fathoms of depth. With a little practice, an operator can adjust the synchronism of the two pulses with a precision of about 0.01 second, which corresponds to an absolute error of about 4 fathoms in the sounding. In sounding moderate depths, this absolute error can be attained in practice, while at great depths the error increases several fold since the returning echo becomes less clearly defined.

The Sonic Depth Finder is essentially a deep-sea device covering the range from the point where the angle of reflection method fails up to several thousand fathoms under favorable conditions. When it is recalled that a single deep-sea sounding by line consumes hours, while the vessel in the meantime drifts about with wind and current, a device of this sort, permitting a continuous line of instantaneous soundings to be taken with the vessel underway, proves of inestimable value in hydrographic surveyings on the high seas and in the laying of trans-oceanic cables. By these means, extensive areas of the ocean bed off the Pacific Coast have recently been charted more thoroughly in a few weeks time than could hitherto have been accomplished in many months of arduous and costly labor.

Another depth-echo device, called the Fathometer, has been contributed by the Submarine Signal Company. In this instrument, a shaft rotating at a fixed constant speed carries an arm which, in one position, on each rotation sends out a short, quickly damped pulse from a submerged diaphragm struck by an electric-hammer mechanism. The returning echo picked up by a microphone system is amplified and subsequently flashes a neon lamp carried on the rotating arm. The angular displacement of this arm from the keying position at the instant of the flash gives, of course, a measure of the depth. The operator merely reads the position of a flashing line of light behind a translucent scale which is calibrated in fathoms. The navigating officer can make these readings with a precision of about one-half a fathom. At depths much greater than 100 fathoms where the echoes become too weak to operate the visual signal, the operator must listen to the echo and note the corresponding depth on the scale as marked off by a continuously illuminated light line carried by the rotating arm, thus judging simultaneity between sound and sight, while, with the Sonic Depth Finder he judges simultaneity between two separate sounds.

Another application of subaqueous acoustics is that of preventing collision of ships at sea in thick weather. In coastal waters, bearings may be taken with modern hydrophone equipment upon the propellers of other vessels up to several thousand yards, thus greatly reducing the chances of collision. On the high seas, however, propeller sounds are useless for this purpose, since the bottom reflected ray becomes too greatly attenuated, due to distance and poor reflection at normal incidence. With higher-pitched sounds from a powerful submarine oscillator, however, the cancellation of the direct and surface-reflected rays

becomes less complete while the bottom-reflected ray is more intense, so that bearings upon such sounds may be obtained at considerable distances. We can picture the ship of the future, when navigating in thick weather, to be transmitting at intervals oscillator code signals giving perhaps the identity and the compass course of the vessel, and sounding a warning to all other vessels which chance to be within audible range. Such subaqueous fog signals would, no doubt, prove much more reliable than the fog horn signals now required by International Marine Law.

Recent experiments of the French and of the U. S. Navy in the field of supersonics, that is, sounds of ultra-audible frequency, offer considerable promise in marine applications. Utilizing the piezo-electric expansion and contraction of quartz crystals, a heavy submerged diaphragm may be vibrated so as to transmit into the sea impulses having a frequency within the range of 20 to 100 kilocycles. Since the wavelength of these sounds is small compared to the size of the radiating diaphragm, it is possible to confine them within a slightly diverging beam, which "pencil of sound" may be swung around the horizon merely by rotating the transmitting diaphragm. If these supersonic waves happen to strike the submerged portion of an iceberg, sufficient energy will be reflected back to be received by a device similar to the transmitter and subsequently amplified and detected audibly by heterodyne methods. Thus, we may anticipate having in the future the supersonic "search light" for locating in thick weather such menacing obstacles as the huge icebergs which in the late spring season threaten shipping on northern routes.

Utopia has not been reached in the hydrophone art. The various devices which have been described are by no means perfected. Much interesting research along many varied lines remains to be done in this field. However, with the submarine bell and oscillator, the multi-spot hydrophone system, the sonic depth finder or the fathometer, and the possibilities of supersonics, it is evident that the science of acoustics has already contributed much towards the safeguarding of lives and property on the seas.

In conclusion I wish to express my appreciation to the Navy Department for their courtesy in permitting me to present this material to you.

SOUND CONCENTRATOR FOR MICROPHONES

HARRY F. OLSON, R.C.A. *Photophone* AND IRVING WOLFF, R.C.A.-*Victor*

Horns have been used for years in conjunction with apparatus for the production and the reception of sound. A horn of practically any shape will cause the ultimate impedance of the receiver end to be obtained at a lower frequency. In the analysis which leads to this conclusion certain assumptions must be made to simplify the theoretical consideration. If the diameter is assumed to be a relatively simple function of the length and the phase remains unchanged over any plane perpendicular to the axis the solution is not difficult. The first of these conditions can be satisfied by choosing the proper shape and the second is satisfied for practical purposes if the diameter is small compared to the wavelength. In practically all horns designed to cover a large frequency band the latter condition is not satisfied over the entire length of the horn for the higher frequencies. When horns of this type are used as receivers of sound the direction of the pencils of high frequency sounds reflected from the large end depends upon the shape of the walls. In general these reflected pencils of sound are not parallel to the axis and therefore considerable change of phase over a plane normal to the axis occurs. The net result is a loss in amplification.

The use of reflectors for the detection and amplification of sound is well known. The surface of the reflector being of such shape that the incident pencils of sound are reflected to one point, the focus. If the surfaces of the reflector is large, and provided the various pencils of sound arrive at this point in phase, considerable increase in pressure over that in the free wave is obtained. In general to realize an appreciable gain in pressure the reflector must be large compared to the wavelength of the incident sound. If this condition is fulfilled for low frequency sounds the size of the reflector becomes prohibitive to be used with facility.

By proper design of the surface of the reflector the gain in sound pickup at low frequencies can be accomplished by action as commonly attributed to horns and at the same time preserve reflector action at the high frequencies. It is the purpose of this paper to describe a parabolic horn designed to function as a horn at low frequencies and a reflector at high frequencies. To this apparatus we have applied the term microphone sound concentrator inasmuch as it is designed to increase

the sound pressure delivered to a microphone by the action outlined above.

Theoretical considerations of horns have been made by Webster,[1] Stewart,[2] Goldsmith and Minton,[3] Slepian and Hanna,[4] Ballantine[5] and others. The theory used to predict the performance of this concentrator in the region where it functions as a horn is the same as used by these investigators.

$$\frac{\partial^2\phi}{dx^2}+\frac{d}{dx}\log\sigma\frac{\partial\phi}{\partial x}-\frac{1}{a^2}\frac{\partial^2\phi}{\partial t^2}=0$$

ϕ is the velocity potential
σ the area of cross section at the point x, and
a the velocity of sound.

The area of cross section of the parabolic horn at the point x is $\sigma=\pi y^2=gx$. The axis of the paraboloid of revolution coincides with the x axis and the vertex is at $x=0$. The general horn equation for the parabolic horn becomes:

$$\frac{\partial\phi^2}{\partial x^2}+\frac{1}{x}\frac{\partial\phi}{\partial x}-\frac{1}{a^2}\frac{\partial^2\phi}{\partial t^2}=0.$$

The solution of this equation is:

$$\phi=[AJ_0(Kx)+BY_0(Kx)]e^{i\omega t}.$$

The pressure and velocity can be determined from this equation for any point along the horn. The length of this horn is determined by planes normal to x axis cutting the paraboloid of revolution at the points x_1 and x_2 the portion of the paraboloid between these two points constitutes the horn.

Let Z_0 represent the acoustic impedance of the opening.

Z_1 represent the acoustic impedance at the small end.

We may consider

$$Z_0=\left(\frac{\rho\omega K}{2\pi}+i\frac{\rho\omega}{C_0}\right)$$

[1] Webster, Journal of the National Academy Sciences, pp. 275–282, 1919.
[2] Stewart, Physical Review, pp. 313–326, 1920.
[3] Goldsmith and Minton, Proceedings Inst. Radio Engineers, p. 423, 1924.
[4] Slepian and Hanna, Journal American Inst. of Electrical Engineers, 1924.
[5] Ballantine, Journal of the Franklin Inst., pp. 85–101, 1927.

(Rayleigh Vol. 2, p. 197–200) where ρ = density of air:

$\omega = 2\pi f$; f = the frequency;

$K = 2\pi/\lambda$; λ = wavelength of sound;

$C_0 = 4R$; R = radius of opening.

The horn is closed at the small end by the microphone and we may assume that:

$$Z_1 = \infty$$

With this information it is possible to find the ratio of the pressure p_1 at 1 and the pressure p_3 in free space. This is given by the expression:

$$\frac{p_1}{p_3} = \frac{[-J_0(Kx_1)Y_1(Kx_1) + J_1(Kx_1)Y_0(Kx_1)}{[J_1(Kx_1)Y_1(Kx_2) - Y_1(Kx_1)J_1(Kx_2)]\left[\frac{\rho\omega K}{2\pi} + \frac{i\rho\omega}{C_0}\right]}$$

$$\frac{}{-J_0(Kx_2)Y_1(Kx_1) + J_1(Kx_1)Y_0(Kx_2) - \frac{\frac{\sigma_2}{i\omega\rho}}{K}}$$

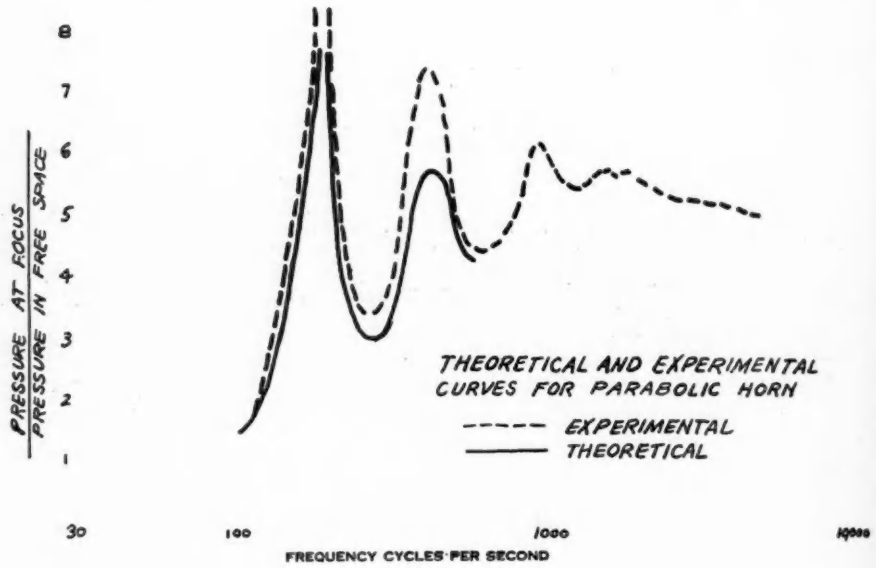


Fig. 1.

The ratio of p_1 to p_3 as derived from this equation is plotted in Fig. 1. The experimental curve was obtained by measuring the pressure at

the focus by means of a condenser microphone and comparing this to the pressure in the free wave at the same distance from the source. An electrodynamic loud-speaker actuated by an audio frequency oscillator was used as a source of sound in these experiments. It will be seen that the very good agreement between the theoretically predicted and experimentally determined performance is obtained. This theory cannot be applied beyond approximately 700 cycles. At this frequency the diameter of the horn becomes comparable to the wavelength of sound. It is possible, however, to show experimentally where reflection by the walls plays an important part.

To test the relative importance of the effects of reflection by the walls upon the performance of the horn a conical section was inserted into the

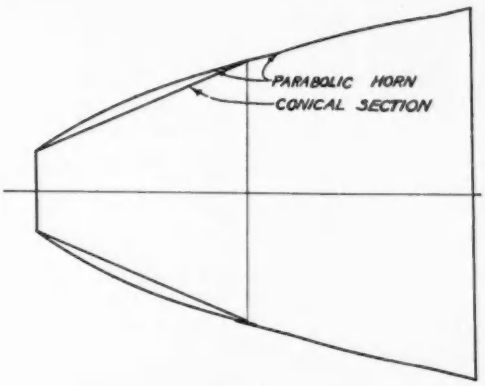


Fig. 2.

parabolic horn as shown in Fig. 2. This divides the horn into two conical horns in series inasmuch as the remaining section of the parabolic horn is approximately a section of a cone. The insertion of this cone has not materially changed the cross-section at any point and the horn action should not differ to any large extent from that of the parabolic horn. In Fig 3 the ratio of the pressure at the small end to that in free space has been plotted from experimental observations made upon the parabolic and the double conical horns. It will be seen that the amplification of the two horns is practically the same below 1500 cycles. A further indication of the similarity is evidenced by almost identical resonant frequencies. These facts show that the action at the lower frequencies has not been upset by insertion of the conical section. Above 1500 cycles the amplification of the double conical horn falls off with increase of frequency whereas the amplification of the parabolic

horn remains practically constant. This indicates that the shape of the walls plays an important part in the resultant amplification at the high frequencies. In the parabolic horn, the pencils of sound arriving from a distant source, situated on the axis, strike the walls and are reflected to the focus in phase, with the exception of the small bundle in the center of the paraboloid, which strike the microphone directly. In the case of the double conical horn the various pencils of sound are not reflected to the same point on the axis and the phase relations of the various pencils at the point of intersection differ depending upon the point of reflection and the frequency. The diffuse focus of the double conical horn

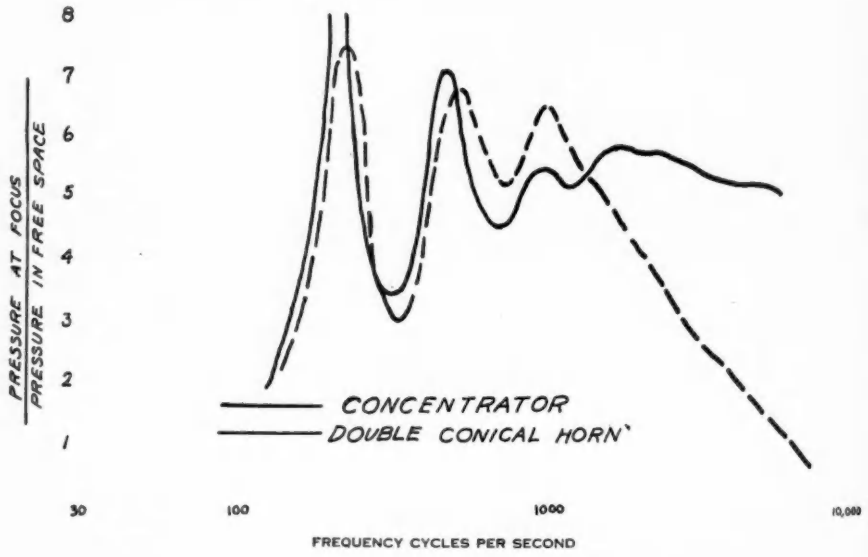


Fig. 3.

results in a loss of the amplification due to lack of concentration. The different phase relations existing between the various pencils at the points of intersection results in a destruction of amplification which becomes more pronounced with frequency. These facts explain in a general way the experimental results of this experiment and we may conclude that reflection from the walls plays a part increasing in importance with the decrease of wavelength of the impinging sound.

The directional characteristics of sound pickup apparatus indicates the solid angle over which sound can be detected without appreciable attenuation of amplification. To investigate the directional properties

of the concentrator, a source of sound, Fig. 4, was placed at a constant distance from the focus and the angle between the axis of the concentrator and the line joining the focus and sound source varied.

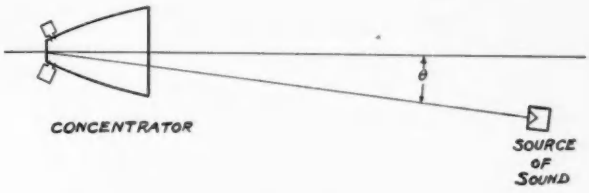


Fig. 4.

The experimental results are shown in Fig. 5. It will be seen that the concentrator shows marked directional characteristics particularly at the high frequencies. In the region where the concentrator functions as a reflector and the sound source is not on the axis, geometrically dissimilar pencils of sound will intersect at a point removed from the focus and in general will differ in phase at the point of intersection. These two

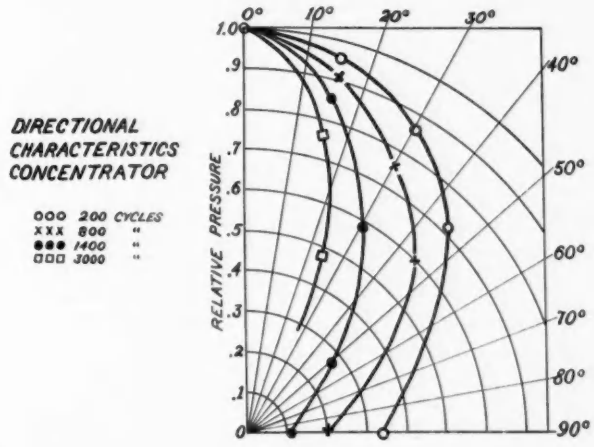
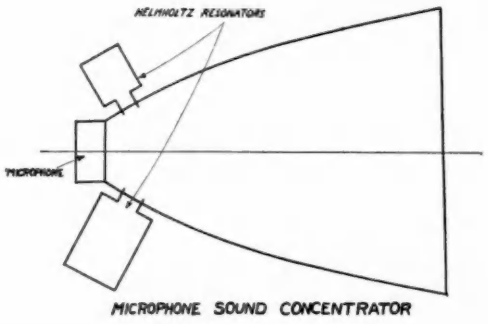


Fig. 5.

factors contribute to a loss of amplification which will become more pronounced with the angle and the frequency. In the region where the concentrator functions as a horn loss in amplification as the sound source recedes from the axis is due to difference of phase existing over a plane normal to the axis which ultimately results in destructive interference.

In any apparatus connected with the reproduction of sound uniform response over the frequency range is desirable and necessary for fidelity of reproduction. The resonance peaks as shown in figure 1 are objectionable and must be brought down either by acoustical or electrical method before this type of concentrator can be employed with sound pickup apparatus. A set of Helmholtz resonators of appropriate design coupled to the horn at the small end has been found very useful for this purpose. A branch consisting of a Helmholtz resonator on a sound transmission line, as shown by Stewart,[6] produces high attenuation at the resonance frequency and considerable attenuation over a wide range bordering the resonance frequency. The action here is complicated by the termination of the acoustic line at the microphone, the point of attachment of the resonator, and also by the fact that the resonator is loosely coupled to



MICROPHONE SOUND CONCENTRATOR

the system. For example, it is found that increasing the diameter of the mouth of the resonator increases the attenuation produced. In this concentrator the diameter is so large at the point of attachment of the resonator, that the action of the resonator is not transmitted over the entire area around the focus. These factors contribute toward incomplete attenuation at the resonance frequency of the Helmholtz resonators. This is, however, the desired result. This type of concentrator in conjunction with the Helmholtz resonators of appropriate design gives an amplification characteristic without serious peaks over the range from 120 to 6000 cycles as shown in Fig. 6.

Although this sound concentrator does not produce a large amplification the gain in sound pressure delivered to the microphone contributes in a large measure to overcoming microphone and amplifier noises.

[6] Stewart, Physical Review pp. 487–492, 1926.

This is particularly useful in outdoor work where the sound reaching the microphone is of low intensity.

The directional properties of sound pickup apparatus determines the ratio of the direct sound from the source to sound originating in random directions.

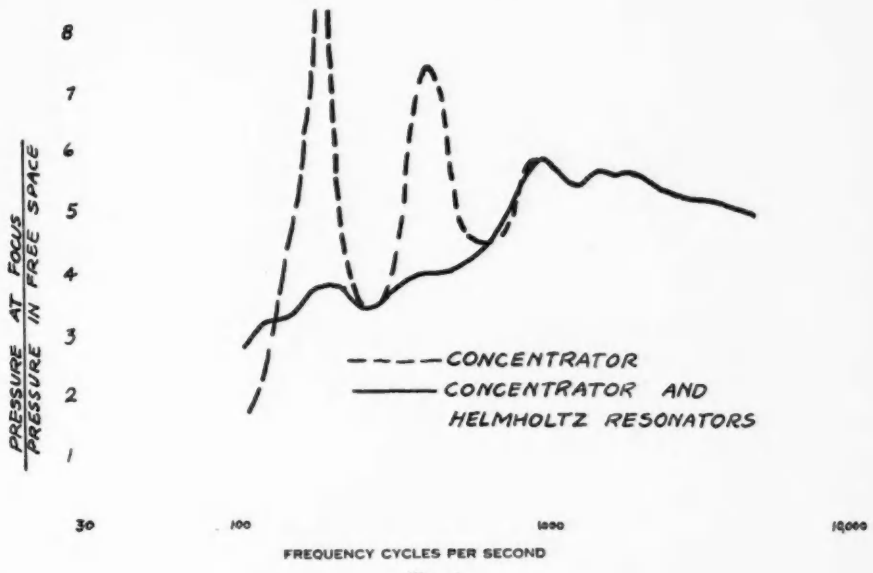


Fig. 6.

This ratio increases as the solid angle over which sound is received without attenuation decreases. When used in rooms or studios the directional characteristics of the apparatus determines the ratio of direct sound to sound originating in random directions. The net result is a reduction in the effects of reverberation. It has been found that the directional characteristics of the sound concentrator contributes in a marked degree in overcoming the detrimental effects of reverberation or generally reflected sounds.

THE MEASUREMENT OF SOUND ABSORPTION BY OSCILLOGRAPH RECORDS

By V. L. Chrisler
Bureau of Standards

This paper describes rather briefly some work which has been carried on at the Bureau of Standards in the attempt to use a recording instrument in place of the ear in measurements of sound absorption. The reverberation method has now proved itself to be the most satisfactory way of measuring sound absorption. Possibly this term "satisfactory" should be modified because of the extent to which the personal element enters into the measurement. For this reason it has been considered desirable to have an instrumental method into which the personal element does not enter and yet which is as near like the reverberation measurement as possible.

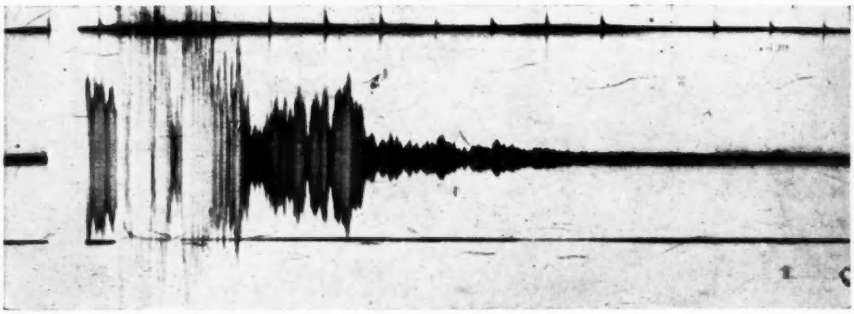


Fig. 1.

One of the various methods that have been tried for this purpose is to use an oscillograph to record the intensity of the diminishing sound. Theoretically the intensity of the sound should decrease logarithmically; actually (as shown in Fig. 1) it does not always do so, at least at the point where it is picked up by the telephone. However, the average intensity throughout the room undoubtedly does decrease logarithmically, and the problem is one of measuring the average intensity.

There are two procedures useful in the attempt to level up the variations in intensity throughout the room. The loud speaker may be rotated or swung through a large arc, and a frequency band may be used instead of a single tone. One of the better films thus obtained is shown

in Fig. 2. In this case it is possible to draw an envelope to the curve and to measure its ordinates at a number of values of the time. Squar-

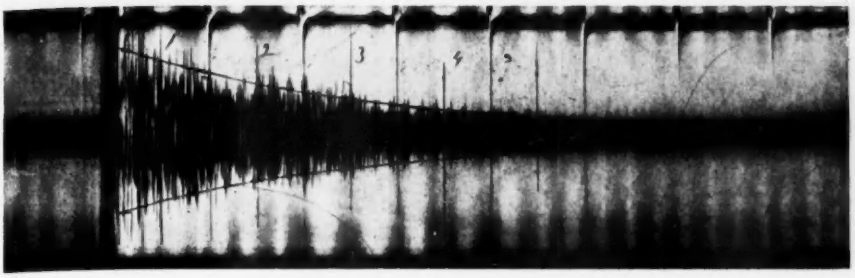


Fig. 2.

ing these ordinates and plotting their logarithms against time the straight line shown in Fig. 3 is obtained. This linear relation may be

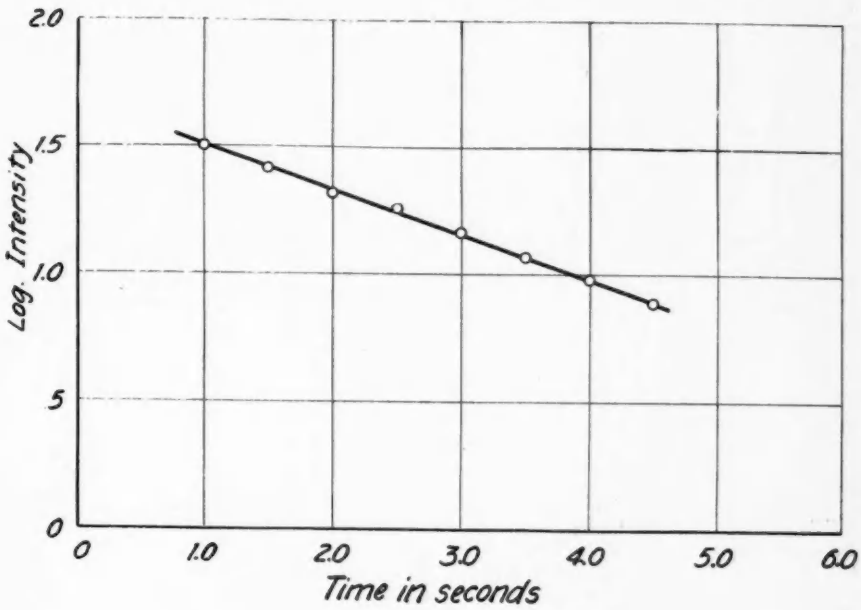


Fig. 3.

expressed mathematically thus:

$$A = K\frac{\log I_2 - \log I_1}{T_2 - T_1}$$

where K is a constant depending on the size of the room and on the velocity of sound. As the formula shows, the fraction is the slope of this line and determines the absorption A.

In this way a calibration of the reverberation room has been made which is shown in Fig.4 along with the calibration made by using several different intensities of sound and determining the reverberation period

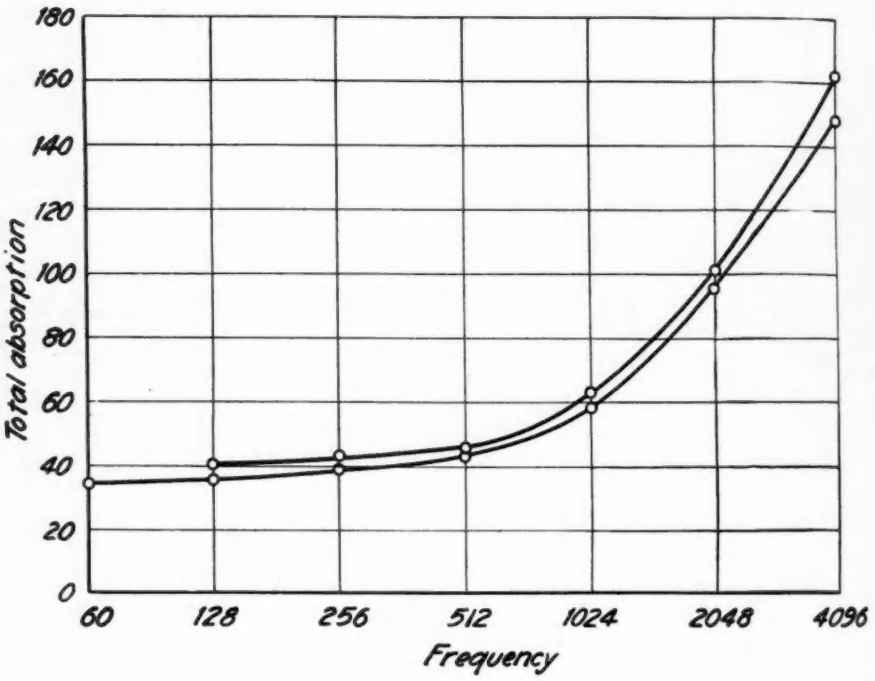


Fig. 4.

by the ear. There was a little additional equipment in the room when the calibration was made electrically but it is doubtful if it would add 4 or 5 units of absorption as indicated. The agreement, however, is good.

An attempt was also made to choke out some of the irregularities by rectifying the current and using some large chokes and condensers in the electric circuit. Fig. 5 shows the result which is not satisfactory for the purpose.

Figs. 3 and 4 show the brightest side of the picture. In many cases the results by the photographic method are greatly in error, sometimes too large and sometimes too small. This is probably because

as the sound dies away there is a gradual drift of the sound pattern in the room past the receiver. It might be considered then that two things are taking place. First the sound is decaying and second this drift of the sound pattern either tends to increase or decrease the rate of decay. If the drift is such as to increase the rate of decay the apparent absorption is too large, while if the drift decreases the rate of decay, the apparent absorption will be too small.

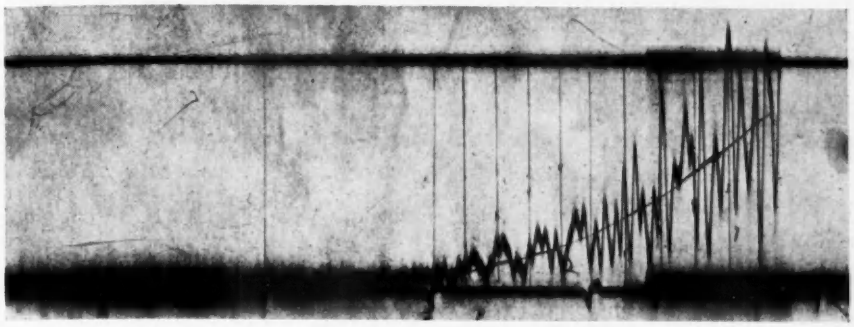


Fig. 5.

If a sufficient number of films were taken this variation would probably average out, as with the ear, but the time and expense involved would be too great. The merit of the ear method does not lie in its precision (for it is not at all accurate) but in the rapidity with which it permits the making of measurements sufficiently numerous to average out their imperfections. The problem still exists, therefore, of finding some way of recording the average decay curve of sound of a single film.

A CHRONOGRAPHIC METHOD OF MEASURING REVERBERATION TIME

By E. C. Wente and E. H. Bedell
Bell Telephone Laboratories

Reverberation time measurements are generally made with the ear and a stop watch in the manner devised by Prof. Wallace Sabine. Surprisingly consistent results can be obtained by this method in a reverberation chamber, where the rate of decay of sound is slow and where disturbing sounds are absent. But such measurements present difficulties if the room is noisy or if the reverberation time is short. Also it is recognized that uncertainties may be introduced because of the fact that the threshold of hearing varies between individuals and with time in the same person. It was with the object of overcoming these difficulties that the electrical method to be described was devised. This method does not differ essentially from that of Sabine except that an

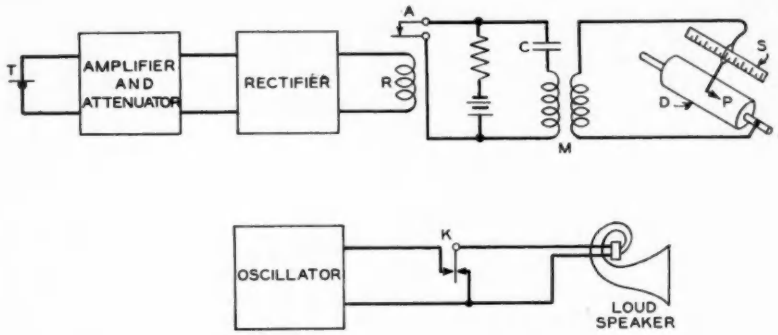


Fig. 1. *Schematic of Apparatus.*

electro-acoustical ear of controllable threshold sensibility is substituted for the human ear.

The general principles of construction of this mechanical ear are indicated in Fig. 1. *T* is a microphone connected to a vacuum tube amplifier provided with an attenuator with which the gain can be controlled in definite logarithmic steps. The amplifier is terminated by a double wave rectifier. The rectified current passes through the receiving winding of a relay, *R*. This relay is so constructed that when the rectified current exceeds a certain value the armature opens the contact at *A*. With this contact open the condenser, *C*, is charged

from the battery, E. When the current falls below a certain value, the armature is released, contact is made at A, and the condenser discharges through the primary winding of the spark coil, M, causing a spark to pass to a rotating drum, D, at P. This spark circuit is similar to that described by C. N. Hickman.[1] The amplifier, rectifier and relay may be so designed that whenever the sound intensity actuating the microphone decreases to a certain value a spark will pass to the drum. This value of intensity may be designated the threshold sensibility of the apparatus.

The drum, D, is rotated at a known, constant speed by an electric or a spring motor. The discharge point may be set at any place along the axis of the drum, the position being indicated on the scale, S. A sheet of waxed paper is placed about the cylindrical surface of the drum, on which a spark discharge leaves a permanent impression.

In the measurement of reverberation time the source of sound may be the usual organ pipe, although a loud speaker driven from an oscillator has generally been used with this apparatus. The tone from the loud speaker can be interrupted by the key, k, which opens the circuit from the oscillator and at almost the same instant short-circuits the speaker. This key is provided with a trigger, not shown, which, after having been set, is released mechanically at a definite angular position of the drum. The time required for the drum to rotate from this position to its position when the spark passes is the time required for the sound to die down to the threshold sensibility of the apparatus.

Measurements of reverbation time are carried out in the following manner. The loud speaker and microphone are placed in the room to be measured. The attenuator is adjusted to a certain value and the spark point is set at some definite position on the scale, S. After the sound has reached a steady state, the trigger of the key, k, is set. This is subsequently released by the rotating drum and the sound from the loud speaker is thereby interrupted. An interval of time later the spark jumps to the drum, leaving an impression on the paper. The gain of the amplifier is then changed by a definite number of *db* and the spark point moved along the scale, S, a corresponding distance; e.g., if the gain of the amplifier is increased by 3 *db* the position of P is moved to the right a half inch and for 6 *db*, one inch. Similarly if the gain is decreased by 3*db*, P is moved a half inch to the left. In this way a series of dots covering a wide attenuation range can be obtained.

[1] Journal of the Acoustical Society, Vol. 1, No. 1, p. 138.

The paper when removed from the drum should then show a succession of dots similar to that shown in Fig. 2. If the decay of the sound at the microphone had been strictly logarithmic these dots would all lie along a straight line. This ideal condition will almost never be encountered in practice. We must therefore be content with drawing a line of best fit through these points. From the slope of the line so drawn and the peripheral speed of the drum, the reverberation time as usually defined is readily obtained.

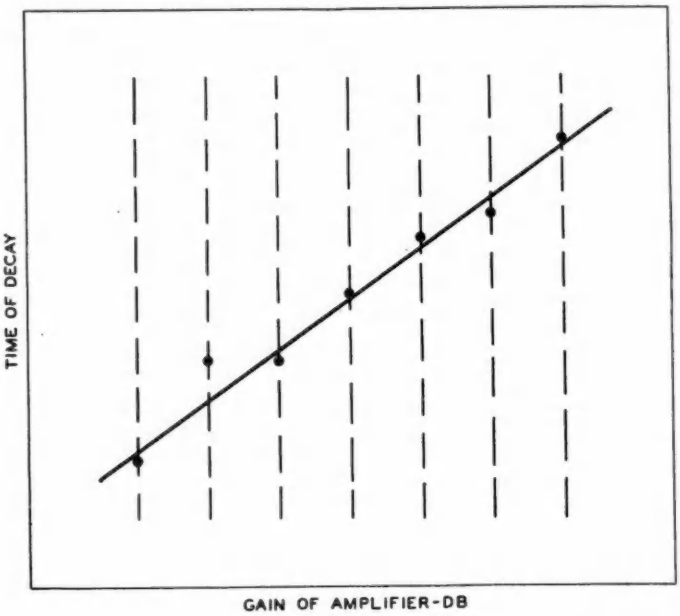


FIG. 2

Due to the shift of the interference pattern as sound dies out in a room the pressure on the microphone will not be a continually decreasing function of the time. In fact, in most cases the energy delivered by the microphone will pass in rapid succession through any previously selected value several times before finally remaining below that value. With the apparatus as described we would get then a group of several sparks as the average energy level passed through the threshold value. Some method must therefore be devised for averaging out the fluctuations in the energy decay curve. The method we have adopted is that of inserting an electrical filter consisting of a single capacitance and inductance between the rectifier tubes and the

relay winding. This filter introduces some time lag in the measurements, the lag being a function of the size of the elements of the filter and of the rectified current through the filter. The inductance and capacitance of the filter should therefore be made as small as is consistent with insuring a single spark for each time interval. The electrical damping in the filter may be made large compared with the acoustical damping in most rooms so that the use of the filter does not affect the measurements; but in order to insure the validity of our results in very dead rooms we have designed the amplifier and rectifier so that the rectifier overloads at a level just comfortably above that necessary to operate the relay. This gives practically a constant current through the filter and relay independent of the amplifier gain, or of the output of the source, over the range of values used. The time lag for each interval measured is very nearly constant therefore, so that the slope of the line obtained is not affected by the delay in the filter.

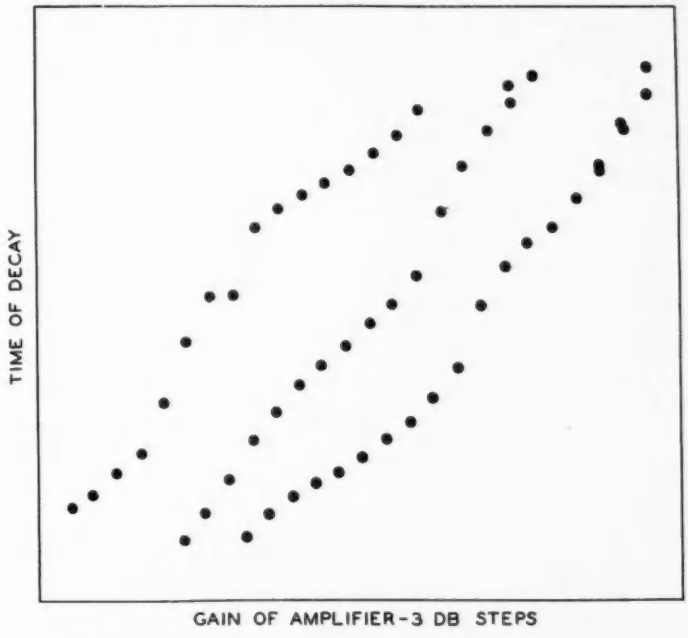


FIG. 3

An indication of the reliability of the measurements may be gained by a comparison of the records obtained with the microphone placed at various positions in the room, other conditions being kept the same.

Fig. 3 shows three such records. Although these records differ considerably from each other, yet, when a straight line is drawn through each set of points, it will be found that the slopes of these lines are nearly the same. These records show, however, the desirability of making measurements over a wide intensity range.

The wavy character of the succession of dots may be much reduced if instead of a pure tone a frequency modulated tone is used.[2] Fig. 4 shows two series of points, the upper one of which was obtained with a pure tone and the lower with a tone having its frequency varied sinu-

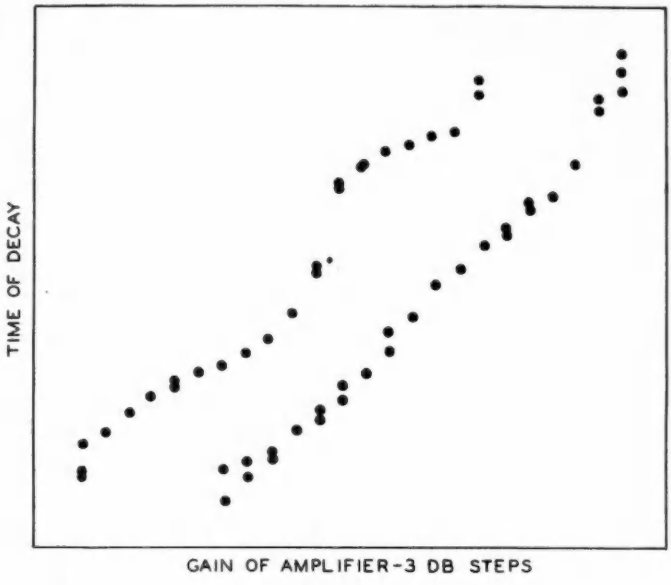


FIG. 4

soidally seven times per second from 1025 to 975 c.p.s. It is seen that in the latter case the points lie much closer along a straight line.

It has also been found a help to place over the microphone a Helmholtz resonator tuned to the measuring frequency. This expedient is particularly helpful in noisy rooms. But even where there is no noise it permits measurements to be made over a wider intensity range, the limit in this case being set by amplifier noise. The time constant of the resonator is, in general, short compared with that of the room, so that no appreciable error is introduced by the use of a resonator, al-

[2] E. Meyer and P. Just, Elektr. Nachr-Techn. 5, 293 (1928).

though at low frequencies in highly damped rooms the damping constant of the resonator may have to be taken into account.

In some of our measurements we have used two microphones with amplifiers and rectifiers of identical characteristics connected in parallel to the relay winding. The microphones were in this case placed in different parts of the room. With this arrangement the dots obtained lie more nearly along a straight line. But except in the case of permanent installations the extra complication introduced can probably not justify itself.

It may be mentioned that instead of varying the gain of the amplifier we may change the input power of the loud speaker and practically the same results will be obtained, unless the loud speaker has a non-linear input-output characteristic.

For successful operation of this apparatus it is necessary that there be no variation in the characteristics of the amplifier and the relay only during the time that one set of points is taken. This condition can be satisfied without difficulty. The amplifier must have a high gain in order to cover a wide range of intensities, but it is not necessary that it be of large power capacity, as no errors will be introduced if the amplifier is overloaded at levels greater than that necessary to operate the relay.

LETTER TO THE EDITOR

February 10, 1930

Professor F. R. Watson
Chairman of the Editorial Committee
Acoustical Society of America
919 North Michigan Avenue
Chicago, Illinois.

My dear Professor Watson:

The Academy of Motion Picture Arts and Sciences has sponsored a series of lectures and demonstrations on the recording and reproducing of sound. During November and December, 1929, four classes were organized, and each class met weekly for a period of ten weeks. The number of students enrolled in these classes was approximately six hundred, and the membership was made up largely of technical workers in the motion picture industry. The course consisted of the following: THE NATURE OF SOUND (two lectures) by Professor A. W. Nye; THE NATURE OF SPEECH AND HEARING by Dr. Vern O. Knudsen; ARCHITECTURAL ACOUSTICS by Dr. Vern O. Knudsen; MOTION PICTURE SOUND RECORDING—PHOTOPHONE SYSTEM by R. H. Townsend, FOX MOVIETONE SYSTEM by E. H. Hansen, WESTERN ELECTRIC SYSTEM by Dr. Donald MacKenzie; FILM AND DISC RECORDING by Colonel Nugent H. Slaughter and Albert W. deSart; WAX RECORDING AND STUDIO ACOUSTICS by J. P. Maxfield; REPRODUCTION IN THE THEATRE by S. K. Wolf and John O. Aalberg; DUBBING SOUND PICTURES by K. F. Morgan; PRACTICAL PROBLEMS IN STUDIO RECORDING by Wesley C. Miller, Douglas Shearer and L. E. Clark.

The Academy plans to commence two additional sections in this course of lectures commencing the first week in March. All of the lectures have been published by the Academy and are bound in a book under the title of "Fundamentals of Sound Recording and Reproduction for Motion Pictures" This bound volume can be obtained through the Academy of Motion Picture Arts and Sciences, Hollywood, California.

Very sincerely yours,
VERN O. KNUDSEN.
University of California at Los Angeles

Theory of Vibration of the Larynx[1]

By R. L. WEGEL

Bell Telephone Laboratories

The vibration in the larynx is caused by an automatic modulation by the vocal cords of the air stream from the lungs. Analytically the mechanism is the same, and physically, closely analogous to that of the vacuum tube oscillator. It depends principally on the resonance of the vocal cords, the modulation of air friction in the glottis by their motion and the attraction due to constriction of the air stream between them. When these forces exist in certain relative proportions and phases, sustained oscillation as in singing takes place. The whole mechanism may be represented analytically by force equations, from which conditions for accretion or subsidence of the vibration or for sustained oscillation may be easily deduced. These equations also show the analogy with other types of oscillating systems.

IT is customary in treating the theory of the voice to assume the glottis or space between the vocal cords to be a source of a steady stream of air with superimposed periodic impulses caused by the vibration of the vocal cords. The harmonic content of these impulses is modified by the "resonating" vocal cavities before being radiated into free air. It is the nature of this modification which receives most attention. The mechanism by which the vibration of the vocal cords is maintained has not been carefully studied.

The vocal cords are maintained in a state of sustained vibration by the proper balance between the various mechanical constants of the complete system, which thus act as a transformer of a part of the non-vibratory power derived from the air stream from the lungs into the vibratory power resulting in sound. It is a simple theory of this mechanism which is considered here.

The method used is to obtain the force equations, which describe the vibrations of the complete mechanical system, by means of the Lagrange equations, from expressions of the total instantaneous kinetic and potential energies, the instantaneous forces acting and rate of dissipation of energy. The resulting simultaneous equations relating to the displacements and velocities of the various parts are then studied to find the frequencies of free vibration and the relations which must obtain between the various mechanical parameters of the system in order that one of these frequencies be sustained. The method is an application of the theory of H. W. Nichols, published in *Physical Review*, August, 1917.

The theory is reduced to easily workable form by the introduction of simplifying approximations which will be described in the progress

[1] Presented before Acoustical Society of America, May 11, 1929.

of the discussion. The principal one of these is the neglecting of all reactions of second or higher order, thus leaving a set of linear differential equations.

Structure of the Vocal Tract

The vocal tract consists of three principal parts, the lungs and associated respiratory muscles for maintaining a flow of air, the

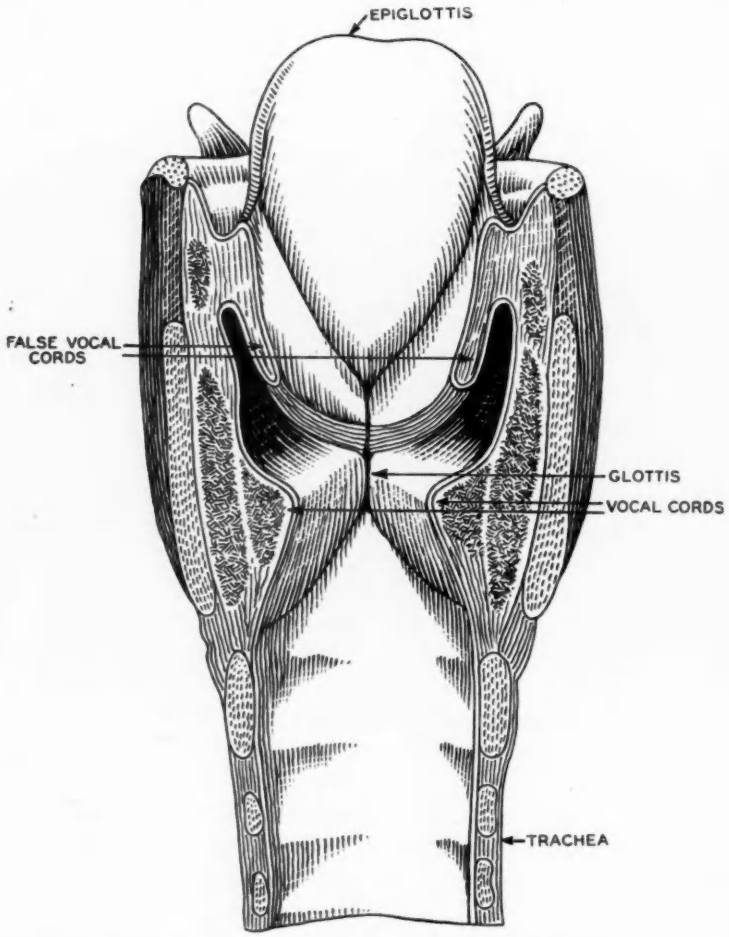


Fig. 1—Anterior-Posterior Section of the Larynx.

larynx (see Fig. 1) for producing the periodic modulation and the upper vocal cavities, pharynx, mouth and nose for varying the relative harmonic content of sound originating in the larynx.

The capacity of the lungs in an adult man is capable of being varied from about two to five liters. The average in quiet breathing is about 2.6 liters. The average expiration of air in quiet breathing is about .5 liter. The rate of expiration of air in medium loud singing varies from 40 to 200 cm.^{3}/sec., the lower values obtaining for trained singers.

The larynx (see Fig. 1) consists of an irregularly shaped cartilaginous box at the top end of a tube, the trachea, about 12 cm. long by 2 cm. in diameter, leading from the lungs. The larynx contains the vocal cords, a pair of fibrous lips which in vibrating vary the width of the slit called the glottis, between them. The length of the glottis in the adult male averages about 1.8 cm. and in the female 1.2 cm. The width of the glottis varies widely with differing sounds. A few tenths of a millimeter may be considered representative. The tension and separation of the vocal cords are controlled by muscles.

The principal upper vocal cavities are the pharynx, a space just over the larynx, the mouth and the nasal cavities. The first and second may be varied in size and shape at will, but the effect on the last is controlled only by varying the communicating aperture between it and the pharynx.

Equations of Motion of the Larynx

Fig. 2 shows a cross-section of a model which illustrates the essential details of the larynx in so far as it is necessary for this treatment. S_0 represents the area of the opening to the trachea. The vocal cords are represented by elastically hinged members of combined effective area S_2. By effective area is meant the area of aperture which displaces the same volume of air as the vocal cords when it moves the distance q_2 of the tips of the cords. This area is less than that of the vocal cords. The tips of the vocal cords are separated to form a gap, the glottis, of area S_1. A positive or up and outward displacement q_2 of the vocal cords increases S_1. It will be assumed that the air is not appreciably compressed in the neighborhood of the glottis, that is, any tendency to compression is relieved by flow into the trachea or pharynx.

The pressure in the lungs forces a steady current of air through the glottis. Let the velocity in the trachea of this steady flow be I_0 and in the glottis I_1. Small vibrations of the vocal cords superimpose additional small velocities, i_0 and i_1, in the trachea and glottis respectively. If the instantaneous velocity of the vocal cords be i_2 and it be assumed that they are constrained to move in synchronism

$$(I_0 + i_0)S_0 = (I_1 + i_1)S_1 + i_2S_2. \qquad (1)$$

The above material is a description of a simple model of two degrees of freedom which simulates the principal characteristics from the standpoint of performance of the more complex larynx which has many degrees of freedom. It is this idealized model which will be considered in the subsequent treatment. Such points of performance of the actual larynx which may be due to the action of ignored and

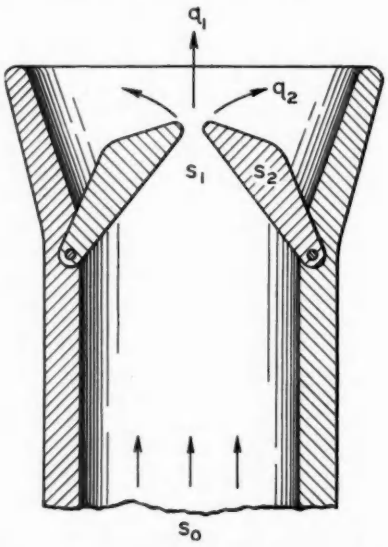


Fig. 2—Schematic Larynx Model.

presumably subsidiary modes of motion will, of course, not be predicted by the theory. These are assumed to be of minor importance. The possible independence of motion of the two vocal cords will be considered later, however.

The contraction of the air stream at the glottis introduces a relatively large concentrated kinetic energy in the air stream at this point similar to that at the mouth of a Helmholtz resonator. The inertia of a small plug of air between the vocal cords may then to a first approximation be treated as a mass L_1. A concentration of frictional resistance also occurs at this point due to viscosity and to turbulence. A positive displacement q_2 (outward) of the vocal cords causes an increase in the mass of the plug of air in the glottis and a change in the effective resistance, R, encountered by it. The inertia L_1' and resistance R of the glottis are therefore both functions of q_2, the displacement of the vocal cords from a mean position, and of the width of the glottis. If further Q_2 represent the average displacement of

the vocal cords from an appropriately chosen reference position, L_2 their inertia coefficient and K_2 their effective stiffness, all measured at the tips, the total kinetic energy, T, and potential energy, V, of the larynx are

$$T = \tfrac{1}{2}L_1'(I_1 + i_1)^2 + \tfrac{1}{2}L_2 i_2^2, \tag{2}$$

$$V = \tfrac{1}{2}K_2(Q_2 + q_2)^2. \tag{3}$$

The Lagrange equation of forces for the nth coordinate of any system is

$$E_n = \frac{\partial}{\partial t}\frac{\partial T}{\partial \dot{i}_n} - \frac{\partial T}{\partial q_n} + F_n + \frac{\partial V}{\partial q_n}, \tag{4}$$

in which F_n is a reaction due to friction. The force equations for the glottis and vocal cords therefore become

$$E_0 = F_1 + \frac{\partial}{\partial t}\frac{\partial}{\partial \dot{i}_1}\left[\tfrac{1}{2}L_1'(I_1 + i_1)^2\right], \tag{5}$$

$$0 = L_2\frac{di_2}{dt} + F_2 + K_2(Q_2 + q_2) - \frac{(I_1 + i_1)^2}{2}\frac{\partial L_1'}{\partial q_2}. \tag{6}$$

Nature of the "Constants" of the System

It is quite safe to conclude that none of the coefficients (inertia, dissipation and stiffness) of the larynx are sensibly constant over the range of operation of the coordinates. Direct measurements are evidently impossible. It is conceivable that they may be arrived at indirectly by means of a comparison of experimental data, especially taken for the purpose, on voice curves and the results of dynamic analysis of the kind described here. The problem may also be studied by means of models. In order to solve equations 5 and 6 it is, however, necessary to evaluate the space and velocity derivatives.

A few simple experiments were performed on models for the sole purpose of determining the qualitative nature of variation of resistance of the glottis with displacement of the vocal cords. A diagram of the model used in the measurements is shown in Fig. 3. This consists of a brass tube, *a*, ¾″ in diameter, beveled off on the top at an angle of 45° with the axis, and two ⅛″ brass plates, *b*, fitted on these beveled surfaces so as to leave a slit, *S*, which was made adjustable in width. A cross-section of this model is shown in *c*. The bottom of the tube was attached to a large air chamber in which the pressure and velocity of air flow could be regulated and measured.

Three shapes of "glottis" were measured. The first had square corners, as shown on Fig. 3*c*. The second, 3*d*, was the same as 3*c*,

except that the corners of the lips were rounded. The third, Fig. 3*e*, had square corners as before, but the slit was about .1 mm. wider in the middle than at the ends.

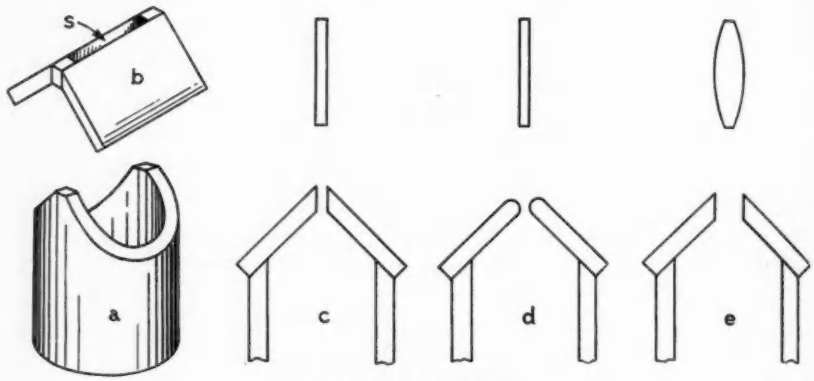


Fig. 3—Glottis Models.

The resistance R is given as the ratio of the product of pressure and slit area to the linear velocity of flow. Measurements were made in each case through a range of pressures such as to give fluxes through the slit through a range of 50 to 200 cm.³/sec. (Stanley and Sheldon values, see *Sci. Am.*, Dec. 1924) and through a range of slit width W of .01 to .10 cm. The data can be represented approximately in this range for the three slits by the following formulæ:

$$R = 3.6\ I^2W^{1.7} \times 10^{-6},$$
$$R = 6.1\ I^2W^{2.2} \times 10^{-6},$$
$$R = 800\ I^2W^{2.3} \times 10^{-6}.$$

In these expressions I is the velocity of flow of air through the slit. More careful data taken through a wider range of I and W would undoubtedly have given R in a power series.

These formulæ are taken to indicate that the resistance of the actual glottis increases faster than a linear function of I and W due to turbulence and may be represented as a single valued function of either displacement of the vocal cords q_2 (or glottis width) or of air velocity as expressed by a Taylor's series as follows:

$$R = R_0 + \frac{\partial_0 R}{\partial q_2} q_2 + \frac{\partial_0 R}{\partial i_1} i_1 + \tfrac{1}{2}\left(\frac{\partial_0^2 R}{\partial q_2^2} q_2^2 + \frac{2\partial_0^2 R}{\partial q_2 \partial i_1} q_2 i_1 + \frac{\partial_0^2 R}{\partial i_1^2} i_1^2\right) + \text{etc.} \quad (7)$$

In this expression R_0 is the resistance measured in the reference position at which point the derivatives are taken, where i_1 and q_2 are zero. The experiment mentioned above determines the signs of the coefficients of q_2 and i_1 as positive. If the flow were purely laminar, i.e. due to viscosity only, the first would be negative and the second zero.

The Reaction F_1

By definition, $E_0 = R_0 I_1$, where E_0 is the force of the lung pressure on the glottis and I_1 a corresponding linear velocity of flow of air. If a force F_1 slightly greater than E_0 act on the glottis and result in a velocity $I = I_1 + i_1$,

$$\frac{F_1}{I_1 + i_1} = R. \tag{8}$$

A combination of (7) and (8) constitutes an evaluation of F_1 for substitution in the force equation (5). To a first order approximation then:

$$F_1 = R_0 I_1 + \left(R_0 + I_1 \frac{\partial_0 R}{\partial i_1} \right) i_1 + I_1 \frac{\partial_0 R}{\partial q_2} q_2. \tag{9}$$

The coefficient of q_2 is dimensionally a stiffness and that of i_1 a resistance. In what follows they will be denoted by

$$F_1 = R_0 I_1 + R_1 i_1 + K_u q_2. \tag{10}$$

Glottis Mass (L_1') Reactions

The kinetic energy of the air stream being proportional to the volume integral of the square of the velocity is largely concentrated in the glottis on account of the relatively high velocity at this point. On account of the irregularity in shape and turbulence in the stream it is impracticable to attempt an integration. If the velocity were so small that the turbulence were absent an approximate value of the air mass would be obtained by taking the mass of a cylinder of air having the length of the slit and a diameter equal to its width. This would make the mass L_1' proportional to W^2, or since the width is proportional to displacement of the vocal cords, to q_2^2. Owing, however, to turbulence and other non-linearities, the mass is probably more nearly described as a tongue of air issuing from the glottis, the inertia L_1' of which varies as some power function of the width and also of the velocity.[2]

[2] It has been found since experimentally that the mass reaction is very nearly that of a cylinder as described but reduced somewhat in diameter due to viscous or turbulent drag at the tips of the vocal cords.

It might be seen by carrying through an expression for this glottis mass involving a function of velocity similar to that for R of equation (7) that only a quantitative change in effective mass would result in the final equations and that no new type of reaction would be introduced. This demonstration is not included here. In order to save space in this qualitative treatment it is ignored. For small displacements q_2 from a reference position at which the velocity of the air is I_0, the glottis inertia may be represented by the direct function:

$$L_1' = L_1 + \frac{\partial_0 L_1}{\partial q_2} q_2 + \tfrac{1}{2} \frac{\partial_0^2 L_1}{\partial q_2^2} q_2^2 + \text{etc.}, \tag{11}$$

in which the coefficient of q_2 is obviously positive. The second term of the second member of (5) may now be evaluated by performing the differentiations as indicated. Neglecting second and higher order terms and denoting dq_2/dt by i_2 the reaction in question becomes

$$L_1 \frac{di_1}{dt} + I_1 \frac{\partial_0 L_1}{\partial q_2} i_2. \tag{12}$$

The glottis mass of air, therefore, introduces two kinds of reactions: a simple inertia and a reaction proportional to the velocity of the vocal cords. For simplicity of notation (12) will be written

$$L_1 \frac{di_1}{dt} + G i_2. \tag{13}$$

This completes the evaluation of the terms (5), the force equation of the glottis, which may now be written

$$E_0 = R_0 I_1 + R_1 i_1 + K_u q_2 + \frac{L_1 di_1}{dt} + G i_2. \tag{14}$$

Force Equation of the Vocal Cords

The force equation (6) of the vocal cords contains four terms. The first is the inertia reactance of the vibrating lips. The mass L_2 is the effective vibrating mass which, if multiplied by one-half the square of the velocity at the cord tip, gives the kinetic energy of their motion. If the distribution of the velocity in the vocal cords were known this might be found by integration. The second term F_2 in equation (6) represents the internal dissipation and is assumed proportional to the small velocity i_2. The third term is the elastic reaction which is proportional to displacement.

The fourth term is a "gyrostatic" term. This term may be written as follows:

$$-\frac{\partial T}{\partial q_2} = -\tfrac{1}{2}(I_1 + i_1)^2\left(\frac{\partial_0 L_1}{\partial q_2} + \frac{\partial_0^2 L_1}{\partial {q_2}^2} q_2 + \text{etc.}\right). \qquad (15)$$

Again by neglecting second and higher order effects this reaction becomes

$$-\frac{{I_1}^2}{2}\frac{\partial_0 L_1}{\partial q_2} - I_1\frac{\partial_0 L_1}{\partial q_2} i_1 - \frac{{I_1}^2}{2}\frac{\partial_0^2 L_1}{\partial {q_2}^2} q_2. \qquad (16)$$

It will be seen that the first term of this expression represents a static force tending, since it is negative, to draw the vocal cords together. This is the Bernoulli effect utilized in a venturi meter. This steady force is counterbalanced by an elastic reaction of the vocal cords with which it combines to determine an equilibrium position which obtains when the cords are not vibrating. This term may, therefore, be dropped from the final equations representing only superimposed motions.

The coefficient of i_1 is identical, except for a sign, with G of (13). It represents a force on the vocal cords due to a superimposed part of the Bernoulli effect caused by the small superimposed velocity i_1 in the glottis. The coefficient of q_2 is dimensionally a stiffness. This apparent stiffness is due to the nature of the air flow and is independent of any elastic members. It is negative if the second differential of glottis mass with respect to cord displacement is negative, positive when this coefficient is positive and vanishes when this coefficient is zero. It simply adds or subtracts in effect from the stiffness K_2 of the vocal cords. The first possibility is the more likely.[3] These terms may then be written for simplicity

$$-\frac{\partial T}{\partial q_2} = -F - Gi_1 - K_x q_2. \qquad (17)$$

Force Equations of the Larynx

The force equations of the glottis and vocal cords with constants thus evaluated are

$$E_0 = L_1\frac{di_1}{dt} + R_1 i_1 + Gi_2 + R_0 I_1 + K_u q_2, \qquad (18)$$

$$0 = L_2\frac{di_2}{dt} + R_2 i_2 + K_2 q_2 + K_2 Q_2 - F - Gi_1 - K_x q_2. \qquad (19)$$

As explained before, $E_0 = R_0 I_1$ and $F = K_2 Q_2$; so these cancel and are of no interest here. In the following it will be seen that the field

[3] This coefficient has since been found to be negative.

stiffness K_x is included in K_2 to simplify notation. This leaves (18) and (19) finally:

$$0 = L_1 \frac{di_1}{dt} + R_1 i_1 + G i_2 + K_u q_2, \tag{20}$$

$$0 = L_2 \frac{di_2}{dt} + R_2 i_2 + K_2 q_2 - G i_1. \tag{21}$$

It should be noted that these equations represent all first order internal reactions of the idealized model of the larynx. The series expansions have been carried out, to show to what approximations these equations hold. It should also be pointed out that the effects of mechanical hysteresis of the parts, which make the relative positions of the parts dependent on the previous history of their motion, have not been considered. A consideration of hysteresis complicates the theory considerably and is ignored for the same reason and with the same justification and limitations that it is ignored in the elementary treatment of electrical circuits containing coils with magnetic material and condensers with electrostatic hysteresis.

External Reactions of the Trachea and Vocal Cavities on the Larynx

So far the modifying effect of the trachea and lungs, as well as the upper vocal cavities, on the motion have not been considered. Before using the equations it is necessary to evaluate these reactions and add them in their proper places.

Imagine a weightless piston fitted into the trachea just below the vocal cords such that the volume of air thus enclosed in the larynx is so small in comparison to that of the trachea and lungs that its compressibility may be neglected. If the vocal cords are held rigid and the plug or piston of air in the glottis is forced inward, a reaction in addition to the resistance and inertia of the glottis will be encountered due to the impeding effect of the trachea piston, which impedance is determined by the constants of the lower chambers. If a small force f_0 act on the trachea, causing a small velocity, i_0, and we assume linearity of response $f_0 = Z_0 i_0$ where Z_0 is a constant which may, due to a positive inertia reactance or a stiffness, contributed by air compression in the lungs, involve either a time derivative or integral of displacement. For the present consider it to be a generalized impedance operator. Due to the relative incompressibility of the air in the larynx, the volume displaced by the trachea piston is $i_0 S_0 = i_1 S_1$. Since also the instantaneous pressure inside the larynx

is constant on all its walls, including the surface of the trachea piston $f_0/S_0 = f_1/S_1$. We then have

$$f_1 = Z_0 \frac{S_1^2}{S_0^2} i_1. \tag{22}$$

This reaction due to the trachea must be added to those of the glottis given in (20). In like manner if the effective area of the vocal cords is S_2 a reaction f_2 must be added to their force equation

$$f_2 = Z_0 \frac{S_2^2}{S_0^2} i_2. \tag{23}$$

Due to the steady component of air flow there is a static component of pressure tending to force the cords outward. This is counter to the static Bernoulli term and again, if second order effects of small quantities be neglected, serves only to alter the equilibrium position and may therefore be disregarded here.

When the glottis plug of air is displaced inward a force is exerted on the vocal cords tending to move them outward which is relieved to a certain extent by a yield of the trachea piston. This force on the vocal cords may be shown by reasoning similar to that above to be

$$Z_0 \frac{S_1 S_2}{S_0^2} i_1. \tag{24}$$

Since this part of the system is linear, the reaction between glottis and vocal cords through this channel is reciprocal so a force is exerted on the glottis when the vocal cords are displaced of

$$Z_0 \frac{S_1 S_2}{S_0^2} i_2. \tag{25}$$

It will be noticed that S_1 is a variable because of the variation in width of the glottis while vibrating. The effect of this variation in these terms is obviously second order since i_1 is small and will therefore be neglected.

The reactions of the upper cavities might be similarly added, but they are apparently relatively small and since they are at present not quantitatively known, are disregarded in the general equations because of the increased complexity. Generally, however, Z_0 may be thought of as representing the additive effects of both upper and lower chambers.

The complete force equations of the voice for small vibrations,

taking into account all major external as well as internal reactions, may then be written:

$$0 = L_1\frac{di_1}{dt} + R_1 i_1 + Gi_2 + K_u q_2 + \frac{Z_0 S_1^2}{S_0^2} i_1 + \frac{Z_0 S_1 S_2}{S_0^2} i_2. \quad (26)$$

$$0 = L_2\frac{di_2}{dt} + R_2 i_2 + K_2 q_2 - Gi_1 + \frac{Z_0 S_2^2}{S_0^2} i_2 + \frac{Z_0 S_1 S_2}{S_0^2} i_1. \quad (27)$$

These equations may be put in a somewhat simpler form by virtue of the fact that they are linear differential equations with constant coefficients. In such a case the time differential may be replaced by an algebraic operator p such that $i = pq$, $di/dt = p^2q$, where p is of the dimensions and nature of a frequency

$$0 = \left(p^2L_1 + pR_1 + pZ_0\frac{S_1^2}{S_0^2}\right)q_1 + \left(pG + pZ_0\frac{S_1S_2}{S_0^2} + K_u\right)q_2, \quad (28)$$

$$0 = \left(-pG + pZ_0\frac{S_1S_2}{S_0^2}\right)q_1 + \left(p^2L_2 + pR_2 + K_2 + pZ_0\frac{S_2^2}{S_0^2}\right)q_2. \quad (29)$$

The determinant of this system is (calling $pZ_0 = Y_0$)

$$D = \begin{vmatrix} \left(p^2L_1 + pR_1 + Y_0\frac{S_1^2}{S_0^2}\right)\left(pG + Y_0\frac{S_1S_2}{S_0^2} + K_u\right) \\ \left(-pG + Y_0\frac{S_1S_2}{S_0^2}\right)\left(p^2L_2 + pR_2 + K_2 + Y_0\frac{S_2^2}{S_0^2}\right) \end{vmatrix}. \quad (30)$$

This determinant represents the complete reactions of the larynx and the external effects of communicating air chambers.

If the effects of the air chambers be disregarded the system is represented by placing $Y_0 = 0$, giving the simple form

$$D = \begin{vmatrix} (p^2L_1 + pR_1) & (pG + K_u) \\ -pG & (p^2L_2 + pR_2 + K_2) \end{vmatrix}. \quad (31)$$

Nature of This System

The voice system represented by determinant (3) is very closely analogous to other vibrators, such as the microphone oscillator or door buzzer and the vacuum tube oscillator. The literature on the latter subject is now so extensive that the pointing out of the analogy should make the method of solution for sustained oscillation, as in singing, or for subsidence or accretion of the oscillation, as in speaking, clear to any one familiar with it.

Fig. 4*a* is a schematic diagram of a three-element vacuum tube oscillator circuit known as the "tuned grid" circuit. This is one of many kinds. The transformer coupling between the plate and grid circuit is represented by an auto-transformer. Fig. 4*b* represents the same circuit schematically but with circuit elements only. In this R_2 represents that part of the resistance of the coil which belongs to the grid circuit and any other associated dissipation, L_2 is the inductance of the coil as seen from the grid mesh and K_2 the reciprocal of the combined tuning capacity across the grid and that of the grid-filament. It is the electrical stiffness or elasticity of the grid mesh,

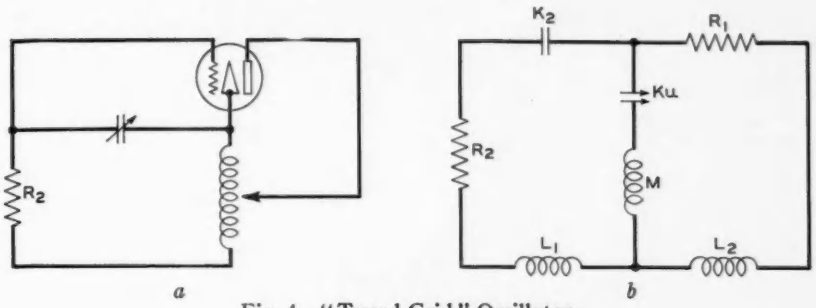


Fig. 4—"Tuned Grid" Oscillator.

in other words. R_1 is a plate-filament resistance ("a.c.") and L_1 that part of the coil in the plate circuit. M is the mutual inductance of the transformer which is not part of the mesh impedance of either plate or grid. The element K_u is the "uni-lateral mutual impedance" (G. A. Campbell, 1914) between the plate and grid meshes and is numerically equal to μK_2, where μ is the amplification constant of the tube. Other internal tube impedances are as usual neglected. The impedance determinant may be written directly from the circuit diagram Fig. 3*b*.

$$D = \begin{vmatrix} (p^2L_1 + pR_1) & (p^2M + K_u) \\ p^2M & (p^2L_2 + pR_1 + K_2) \end{vmatrix}. \qquad (32)$$

The quantities on the principal diagonal of this determinant, that is the first and last elements, are as usual in a circuit determinant the mesh impedances while the others are the mutuals. The principal features of the analogy may be seen by comparison of determinants (31) and (32). Except for the thus far undefined external or trachea impedance the mesh impedances are the same, from which it appears that the glottis is analogous with the plate-filament path in the vacuum tube and the vocal cords with the grid-filament path. The air

velocity in the glottis I_1 corresponds to the plate current. In the vacuum tube this plate current is modulated by varying charge, q_2, on the grid. In the larynx the glottis air velocity is modulated by varying displacement, q_2, of the vocal cords. The charge on the plate (again neglecting internal capacities except the grid-filament) causes no effect on the grid mesh and in the larynx the position of any element of glottis air has no effect on the vocal cords. The unilateral mutual impedance, K_u, is the same in both.

The analogy breaks down at the point where the " feed back " part of the mechanisms is compared. The " feed back " is the bilateral part of the mutual impedance between the two meshes. In the vacuum tube circuit this is p^2M, the mutual of the transformer, while in the larynx it is pG, the " gyrostatic " mutual. The latter is a type of element which does not occur in electrical circuits, arising as it

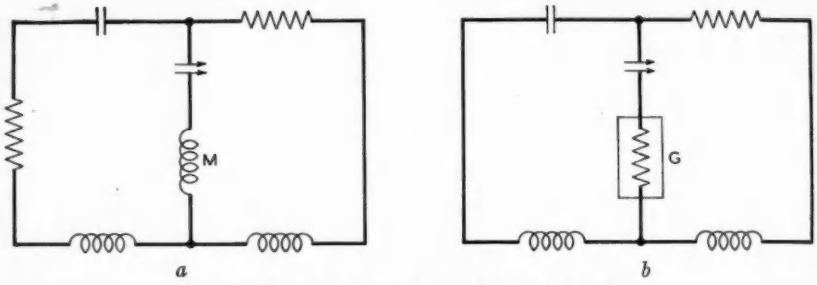


Fig. 5—Tuned Grid and Wind Reed Circuits.

does from a variation of a mass or inductance with a displacement. Inductance, being a function purely of the geometry of a circuit, can only vary with mechanical displacement and not with electrical displacement or charge. The gyrostatic mutual is common in the mechanics of rotating bodies whence it derives its name. It is also the mutual in an electromagnetic telephone receiver or relay between the electrical circuit and the armature or diaphragm.

In order to fix the rather useful concept of the analogy in mind, Fig. 5 is added showing the schematic circuit of the vacuum tube (5*a*) and a circuit diagram (5*b*), which represents determinant (31) the characteristic formulation of the dynamics of the larynx. Fig. 5*b* is represented by the conventions of an electrical circuit, except for the element G for which a different convention is necessary. The one taken here is that of a resistance enclosed in a rectangle. From (31) it will be seen to be similar to a resistance in its association with frequency p but different from resistance in that it occurs non-symmetrically in sign in the determinant. It does not involve dissipation.

Its occurrence here is the simplest possible for when there are appreciable concealed or ignored modes of motion it may have the form of a generalized impedance containing at least one element of resistance, but will always be non-symmetrical as a whole in sign in the determinant.

The use of the circuit for representing the mechanical system is an extension of an old but recently popularized method of studying mechanical or electrical vibrating systems by the help of analogy, one with the other. The extension consists in the explicit representation by diagram of the gyrostatic mutual which makes the determinant unsymmetrical in sign and of the unilateral mutual which makes the determinant unsymmetrical in magnitude. Fig. 6 is a

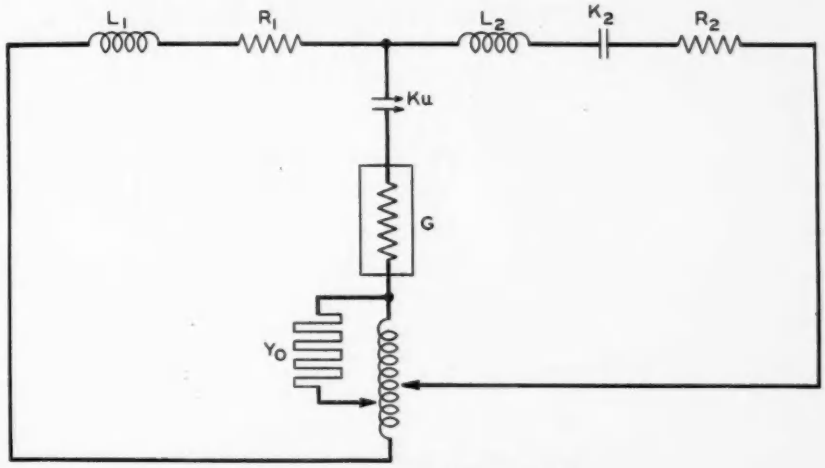


Fig. 6—General Wind Reed Circuit.

diagrammatic representation of the more general system of determinant (30). This includes the external Y_0 reactions as well as the internal.

Having thus described the extended method of analogy the following study of the larynx with the help of the circuit diagram of its determinant should be clear.

Sustained Vibration of the Simple Larynx

In vibrating, the vocal cords do not receive excitation of the frequency at which they vibrate. The source of power is in the air stream I_1 which enters the equations in K_u, the unilateral mutual impedance. Since this is treated as a constant circuit or dynamical element this air stream may be ignored as a drive and the resulting

vibration considered as the free oscillation of the system. The determinant (31) (or 30) is then used to determine the free frequencies and decrements of the system. The method is as usual to solve for p in the equation

$$D = 0. \tag{33}$$

To simplify the demonstration the simple larynx without the load of the air chambers will be considered. Taking D of (31) then and expanding:

$$p^4L_1L_2 + p^3(L_1R_2 + L_2R_1) + p^2(L_1K_2 + R_1R_2 + G^2) + p(R_1K_2 + GK_u) = 0. \tag{34}$$

If this be divided by L_1L_2 and the uncoupled decrements and natural frequency defined:

$$\frac{R_1}{2L_1} = \Delta_1; \quad \frac{R_2}{2L_2} = \Delta_2; \quad \frac{K_2}{L_2} = \omega_2^2, \tag{35}$$

then

$$p\left[p^3 + p^2(2\Delta_1 + 2\Delta_2) + p\left(\omega_2^2 + 4\Delta_1\Delta_2 + \frac{G^2}{L_1L_2}\right) + \left(2\Delta_1\omega_2^2 + \frac{GK_u}{L_1L_2}\right)\right] = 0. \tag{36}$$

One of the roots of this equation is zero and another is negative real since all coefficients are positive. This root is therefore the decrement of a mode of non-vibratory motion. The remaining two roots may be real, imaginary or generally complex, of the form

$$\Delta \pm j\omega. \tag{37}$$

If it is found that $\Delta = 0$, then an oscillation once started will be sustained. If Δ be negative then any existing oscillation must subside or if Δ be found positive then an impulse will start an oscillation which of itself increases in amplitude to a point where its violence modifies the constants to such an extent as to make Δ vanish, leaving a sustained oscillation, or negative leaving the oscillation to subside to a lower amplitude or completely if sufficient permanent changes have been made.

If now (36) be written

$$Ap^3 + Bp^2 + Cp + D = 0 \tag{38}$$

and the first root (37) be substituted, two equations result, one from the real and the other from the imaginary terms, as follows:

$$A\Delta(\Delta^2 - 3\omega^2) + B(\Delta^2 - \omega^2) + C\Delta + D = 0, \tag{39}$$

$$A(3\Delta^2 - \omega^2) + B(2\Delta) + C = 0. \tag{40}$$

Now the condition for sustained oscillation is that $\Delta = 0$ and if the value of ω when this obtains be ω_0 then

$$\omega_0^2 = \frac{D}{B} \qquad \text{and} \qquad \omega_0^2 = \frac{C}{A}, \tag{41}$$

or the condition for sustained vibration in terms of the constants is

$$AD = BC. \tag{42}$$

In addition to this if use be made of the fact that in an algebraic equation such as (36) the coefficient of p^2 is the negative sum of all the roots then this coefficient is the real root. Let this be Δ_0 and then (36) may be written

$$p^3 + p^2\Delta_0 + p\omega_0^2 + \Delta_0\omega_0^2 = 0. \tag{43}$$

The coefficient of p in (36) is therefore the square of radial frequency at which sustained oscillation will take place and this is seen to be higher than the natural frequency ω_2 of the vocal cords, the difference being increased when the damping of either mesh is greater or when the coupling mutual G is greater.

It might be noted in passing that (43) is the free oscillation equation for any system which may be represented by a cubic equation and is not confined to the simple larynx. Such an equation always results when there is only one kind of reactive element in one of the meshes. It holds also for the tuned grid circuit.

The condition for sustained oscillation to be fulfilled for the constants may from (42) be reduced to:

$$R_1R_2 = G^2\left(\frac{L_1K_u}{GR_1} - 1\right). \tag{44}$$

It is rather difficult to place a simple physical interpretation on this formula. The qualitative import of it may however be seen by substituting the values of G and K_u from (13) and (10):

$$R_1R_2 = L_1^2I_1^2\left[\frac{\partial_0 R/\partial q_2}{R_1} - \frac{\partial_0 L_1/\partial q_2}{L_1}\right]\frac{\partial_0 L_1/\partial q_2}{L_1}. \tag{45}$$

The first term in brackets is in the nature of a resistance modulation constant, a fractional change in glottis resistance per unit displacement of the cords, to be designated by r and the second term similarly a glottis mass modulation constant, ℓ. The quantity L_1I_1 is the momentum of the air in the glottis. This equation is then

$$R_1R_2 = (L_1I_1)^2(r - \ell)\ell. \tag{46}$$

Thus it appears that the resistance modulation must always be greater than the mass modulation and when the difference is small the air momentum must be increased to compensate. Owing to the physical limitation in accuracy of continuous maintenance of adjustment in the larynx, if a large momentum is depended upon to compensate for a small modulation difference, an unsteadiness or instability is likely to result. It is common experience that it is impossible to produce a sound with the voice with less than a certain minimum intensity. This corresponds, with the most favorable adjustment of the modulation constants which are physically possible, to a minimum momentum of air from the lungs which satisfies (46). It will be evident that this interpretation must not be taken too seriously quantitatively.

Subsidence and Accretion of Vibration of the Simple Larynx

Oscillograms made of the speaking voice show that, among other things, the amplitude of the oscillation and the pitch are in a continuous state of change. This is also true in singing but not nearly to the same extent. It seems therefore that in singing the adjustment of the voice system for sustained oscillation as described in (44) above is of major importance, while in speaking conditions for variation are of most importance.

The principle of the investigation of variation is simple enough but in all but the most elementary systems the algebra involved is impracticably awkward. If by solving (33) directly for the roots of p, it be found that Δ is positive, then any existing vibration will tend to increase while if Δ is negative, then vibration will tend to subside. The algebraic difficulties arise in the general solution but these are largely obviated by making the assumption, which is most likely usually fulfilled in practice, that the real parts of the roots may be treated as small quantities when compared with the imaginary parts. A common frequency for a man's voice is 150 cycles per second for which ω_0 is 1000 in round numbers. The decrement of a telephone receiver is ordinarily 100 to 200 in open air. The decrement of a tuning fork is represented by a fraction. Judging from variations in amplitude in an oscillogram (from which of course decrements may not be read directly) it would seem reasonable to assume that Δ is small compared with ω_0. The study of variation thus becomes an investigation of small departures from a condition of sustained oscillation, the reference condition being that critical adjustment for which the roots of interest of (33) are pure imaginary.

Suppose in (38) that $A = 1$; then without loss in generality:

$$p^3 + Bp^2 + Cp + D = 0. \tag{47}$$

In such an equation the roots are continuous functions of the coefficients. The same is true of their derivatives except at the one point where transition occurs from pure real to complex values. The values of the roots of interest in this connection are in their complex region at the point where the real part of the root passes through a zero value. This is the point at which free oscillation of the oscillating mode occurs, the values of the roots of this mode being as shown before, $\pm j\omega_0$.

If it now be supposed that one cause produces small variations, directly or indirectly on each of the coefficients and that the magnitude of this cause be x, then:

$$(3p^2 + 2Bp + C)\frac{dp}{dx} + p^2\frac{dB}{dx} + p\frac{dC}{dx} + \frac{dD}{dx} = 0. \tag{48}$$

The problem then is to determine dp resulting from any assigned cause dx when $p = j\omega_0$. From (43) we have at this point $B = \Delta_0$, $C = \omega_0^2$ and $D = \Delta_0\omega_0^2$.

$$2\omega_0^2\left(1 - j\frac{\Delta_0}{\omega_0}\right)\frac{dp}{dx} = -\omega_0^2\frac{dB}{dx} + j\omega_0\frac{dC}{dx} + \frac{dD}{dx}. \tag{49}$$

This is the frequency (complex) variation equation taken in the neighborhood of free oscillation.

When any readjustment of the larynx takes place all of the "constants" entering the coefficients undergo change, in particular those of the glottis K_u, R_1, G. Suppose for simplicity that one only varies, then this variation dK_u, dR_1, or dG may be taken as the magnitude of the cause dx. In particular if K_u vary,

$$dB = 0 = dC \quad \text{and} \quad dD/dx = G/L_1L_2,$$
$$\left(1 - j\frac{\Delta_0}{\omega_0}\right)\frac{dp}{dK_u} = \frac{G}{2\omega_0^2L_1L_2}. \tag{50}$$

If in addition Δ_0 be small compared with ω_0,

$$dp = \frac{GdK_u}{2\omega_0^2L_1L_2}\left(1 + j\frac{\Delta_0}{\omega_0}\right). \tag{51}$$

This shows that if a condition of sustained oscillation is departed from by slightly increasing K_u, an increase in the amplitude of vibration begins which is proportional to the logarithm, since $(p + dp)$

is the exponent, of the increment dK_u and the frequency (imaginary part) of vibration increases slightly in proportion. If K_u were the only varying element the vibration would continue indefinitely to increase.

If on the other hand K_u be assumed constant, the variation being in R_1, then it may be similarly shown that

$$dp = \frac{d\Delta_1}{2\omega_0^2}\left[-\left(4\Delta_1 + \frac{G^2}{L_1L_2} + \Delta_0^2\right) + j\frac{\Delta_0\omega_2^2}{\omega_0^2}\right], \qquad (52)$$

whence it appears that a small increase in glottis resistance dR_1 (or $d\Delta_1$) introduces a subsidence of vibration but an increase in frequency of oscillation as before. A decrease $-dR_1$ of course produces the opposite effect.

If the change be in G, it turns out that

$$dp = \frac{dG}{2\omega_0^2 L_1L_2}\left[\left(K_u - 2G\Delta_0\right) + j\left(\frac{K_u\Delta_0}{\omega_0} + 2\omega_0 G\right)\right]. \qquad (53)$$

Here it appears that an increase in the gyrostatic mutual, G, may introduce either a subsidence or an accretion in amplitude but like the others makes for an increase in frequency of oscillation.

Variation in other elements produces similar conflicting tendencies not only in damping but in frequency.

The physical picture to be drawn from this is that in speaking the voice modulates from one amplitude and frequency to another by proper relative variations in adjustments in its constants, being constantly in a state of changing subsidence or accretion. It would seem also that the principal cause of change in frequency is in the vocal cords and that of amplitude variation in the glottis. Speaking is, in this respect, a more intricate process than singing.

Other Types of "Feed Back"

The detailed study of the larynx has so far been limited to the assumption that the "feed back" is entirely gyrostatic. This is of course actually not the case. How much influence is exerted by the general Y_0 is difficult to estimate.

If the trachea were a long tube but still shorter than a quarter wave-length of sound at the frequency of oscillation and rather smaller in diameter, and substantially open at the end the mass of the air in it would then be appreciable and Y_0 in (30) would be written p^2L_0. If in addition the gyrostatic term were negligible the system would then be exactly analogous with the tuned grid system and (32) rather than (31) should be the subject of detailed study.

If on the other hand the lungs acted substantially as a solid walled chamber of comparatively small size, the elasticity of the contained air would be represented by taking K_0 for Y_0. The surface area in the lungs is very large compared with a regular chamber of equal volume so considerable dissipation must be encountered by vibration. If this were the most important reaction Y_0 should have been replaced by pR_0.

Unquestionably all three types of reaction enter. A more general treatment to include them is plainly not a subject for a short paper. It is interesting however to note that in the dynamical system of brass horns these latter Y_0 reactions exert controlling influences. In this case the lips of the player perform the same function as do the vocal cords of the voice while the external load, the horn, corresponds to the pharynx, the reaction of which is the same dynamically as the trachea. In this case the frequency of the horn is that of sustained oscillation and not that of the lips. The same is true of the wood-wind, in which case the reed or reeds replace the lips or vocal cords. In these cases Y_0 is proportional inversely to the hyperbolic tangent of the frequency or may be approximately represented by the impedance of an anti-resonant element.